Changes to UCR 2008, Change 2, Section 6.1, Unique Capabilities and Requirements

SECTION	CORRECTION	EFFECTIVE DATE
Several sections	Circuit-switched requirements have been removed. Please refer to UCR 2008 for these	Not Applicable
6.1.4	Deployed Network Element requirements have been moved to Section 5.9, Network Elements	Must Be Implemented Immediately as specified in Section 5.9 errata sheet

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SECTION 6
UNIQUE CAPABILITIES AND REQUIREMENTS

Section 6 contains requirements that are unique to Deployed (Tactical) systems in Section 6.1, Unique Deployed (Tactical) Requirements, and classified systems in Section 6.2, Unique Classified Unified Capabilities Requirements. The unique requirements are modifications to, or additions to, the overall requirements defined in Section 5, Unified Capabilities Product Requirements.

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6.1 UNIQUE DEPLOYED (TACTICAL) REQUIREMENTS

6.1.1 Introduction

This section defines unique Deployed (Tactical) requirements. Section 5.1, Requirements Categories and Language, provides the general definition of requirements terminology used. This section was created by the Executive Agent for Theater Joint Tactical Networks (EA-TJTN) to identify the Tactical requirements in consonance with responsibilities assigned by ASD(NII). In addition, the U.S. Military Communications-Electronics Board (MCEB) tasked the Theater Joint Tactical Networks Configuration Control Board (TJTNCCB) to develop Tactical interoperability requirements as certification criteria for joint networked-communications systems. In pursuing acquisition initiatives, COCOMs, military services, and defense agencies shall use this section as a guideline for the purchase of COTS equipment as well as for the development of systems that need to interface in deployed networks. The Tactical networked communications community of the DoD shall adhere to this section to comply with DoDI 8100.04.

6.1.1.1 Purpose

This section defines the unique requirements for Deployed products and systems. These are requirements that are not contained in the Fixed requirements sections of the UCR, and define Fixed requirements that need to be modified to support Deployed users. This section consolidates interoperability certification requirements to the maximum extent possible and incorporates them as part of requirements for the overarching GIG in support of network-centric warfare. This section provides guidance for satisfying the certification requirements for Deployed voice systems used as part of an OAN, which is the deployed extension of the GIG. This section also defines other UCR elements applicable to the Deployed community, and serves as a ready reference to be used by the JITC when writing the Deployed annex to the Generic Switch Test Plan (GSTP).

6.1.1.2 Applicability

The requirements described in this section apply to NEs, LANs when used in Deployed (Tactical) environments, Deployed Cellular Voice Exchange (DCVX) Systems, and LSCs.

6.1.1.3 Definitions

Definitions and acronyms are provided in Appendix A, Definitions, Abbreviations and Acronyms, and References.

6.1.2 Circuit-Switched-Based Deployable Network Designs and Components

Circuit-switched-based deployable requirements defined by previous editions of the UCR remain in effect during the remaining life cycle of deployed circuit-switched products.

6.1.3 Deployed Voice Quality

The desired objective for Deployed voice quality is an MOS of 4.0 or greater, but it is realized that the network may operate under less than ideal conditions. The requirements provided in the following paragraphs are the minimally acceptable values under the conditions specified. The MOS calculation will assume the use of G.729 with 20 ms samples for the purpose of SLAs.

6.1.4 Deployed NE General Requirements

Section 5.9, Network Element Requirements, contains the Deployed NE general requirements.

6.1.5 Deployed LANs

6.1.5.1 Overview

Tactical Operations Centers (TOCs) and other deployed enclaves operate under austere conditions, rely on a Deployed power supply or grid, and may be restrictive in the size, weight, and packing requirements. The Deployed LAN and the backbone and transmission components operate from the same Deployed power source. It is extremely difficult to approach the availability and power backup requirements mandated on the Fixed infrastructure with its commercial-grade power supply and fixed operating environment.

The ASLAN requirements defined in Section 5.3.1, Assured Services Local Area Network Infrastructure, represent the optimal LAN design. Deployed users are encouraged to implement their requirements whenever possible. However, operational realities often preclude the deployment of highly redundant components and multiple backup power sources.

6.1.6 DCVX System Requirements

6.1.6.1 Introduction and Purpose

The following sections describe the requirements that shall be met by all deployed DCVX systems to be certified and used in the OAN tier of the Global GIG. Requirements are defined at the system level as well as for the various components that make up the cellular system, including protocol requirements. The DCVX is a cellular system with military-unique features (MUFs), and therefore, is not the same as commercially deployed cellular systems.

It is recognized that not all components are needed for a specific application. The requirements discussed in this section are similar to those for a Deployed Voice Exchange-Commercial (DVX-C) and/or Local Session Controller (LSC), and are dependent on the network configuration as well as the specific authorized gateway connection.

6.1.6.2 Applicability

The requirements within this section are applicable to the following:

- All DCVX systems that connect directly or indirectly to the DISN voice systems, including the UC Services Network, DSN, DRSN Secure Phone Gateways, and/or commercial PSTN.
- Procured or leased commercial cellular systems that connect to any DISN service gateway. Commercial cellular services are not currently allowed to be directly connected to DISN service gateways unless the connection is TDM-based (e.g., Analog, PRI, or ISDN), excluding the use of SS7. Future commercial cellular services' IP-based connections will be allowed once the Information Assurance policy and STIGs are established. In both instances, the DISN service gateway may or may not be protected by a separate or built-in encrypted gateway on the commercial cellular services connection. Encrypted gateway requirements are excluded from the DCVX section.
- Procured or leased cellular systems using leased commercial cellular frequencies that connect to any DISN service gateway.

Terminal devices procured and/or leased, whose primary carrier service is owned and operated solely by a commercial carrier service (e.g., Verizon, Sprint, etc.) are not considered elements of a DCVX and are exempt from this section. These types of terminal devices are detailed in Section 5.4.5.4.9, Smartphones, Mobile UC Applications, and Backend Support Systems. The current version of the UCR is the governing requirements document that takes precedence over the explicit or implicit requirements of subsidiary or reference documents, standards, and specifications. In the event of a conflict, the explicit requirements of the UCR take precedence over the explicit or implicit requirements of any other requirements document except for those requirements specified in the documents listed in Section 6.1.6.3, Policy and Reference Documents.

6.1.6.3 Policy and Reference Documents

The following policy and instruction documents, in conjunction with the current version of the UCR, will be used as the basis for APL certification:

1. Policy for the use of commercial wireless devices, services, and technologies in the DoD GIG, as outlined in DoDD 8100.2. This directive further promotes joint interoperability using open standards throughout DoD for commercial wireless services, devices, and technological implementations.

2. "Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for CDMA-Based Systems – Home Location Register (HLR)" or current edition.

3. "Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for GSM-Based Systems" or current edition.

4. 3G TS 24.067 V3.0.0 (1999-05), 3rd Generation Partnership Project; Technical Specification Group Core Network; enhanced MLPP (eMLPP) – Stage 3 or current edition.

6.1.6.4 DCVX System Overview

The DCVX systems provide wireless mobile communication services with MUFs and draw their Strategic services by approved DoD authorized gateway switching systems only. The DCVX can be connected to a DVX-C, connected directly to the DSN, and/or to the UC Services Network utilizing AS-SIP for TDM and IP switching systems, respectively. The DCVX systems also may be interconnected with other cellular telephone systems, excluding commercial systems, unless the commercial system is procured or leased for DoD usage and is operating in an isolated mode from other commercial provider cellular systems.

When placed in a Deployed environment, the DCVX will have the capability to connect to DSN/UC Services and between other DCVXs and DVX-Cs using UCR-defined protocols such as ISDN PRI, MLPP PRI (T1.619a), and/or AS-SIP. A DCVX system may also be configured to interconnect at the network transmission level with other DCVX systems to provide roaming capability outside the local home base cellular network for supported terminal devices. In support of this roaming capability, the DCVX cellular systems may interconnect based on the interconnection protocol requirements of the appropriate 2G, 3G, and/or 4G standards.

The DCVX terminal devices, often referred to as mobile subscriber cellular handsets, PDAs, Smartphones, BlackBerry®s, and any other end user cellular devices, commercial or Government developed, may connect to commercial cellular systems when operating outside the transmission range of the DCVX. Additionally, the cellular terminal devices may have the capability to interface with other wireless networks (e.g., IEEE 802.11 and IEEE 802.16).

Actual employment of this additional cellular terminal device capability will be by command approval only in the Tactical OAN.

6.1.6.4.1 DCVX Components

The DCVX is comprised of the following three major functional areas: Terminal devices(s), Access Network, and Core Network. Terminal devices can be mobile subscribers' cellular handsets, PDAs, Smartphones, BlackBerry, or any other end user cellular devices, commercial- or Government-developed. With the evolution of cellular technology from 2G to 4G, the primary functional components that comprise the DCVX Access and Core Networks are evolving as well. For comparison of the primary functional Access and Core Network components that comprise an operational DCVX across the evolutionary changes, Figures 6.1.6-1 through 6.1.6-3 provide the primary cellular architecture Access and Core Network components for 2G, 3G, and 4G systems, respectively.

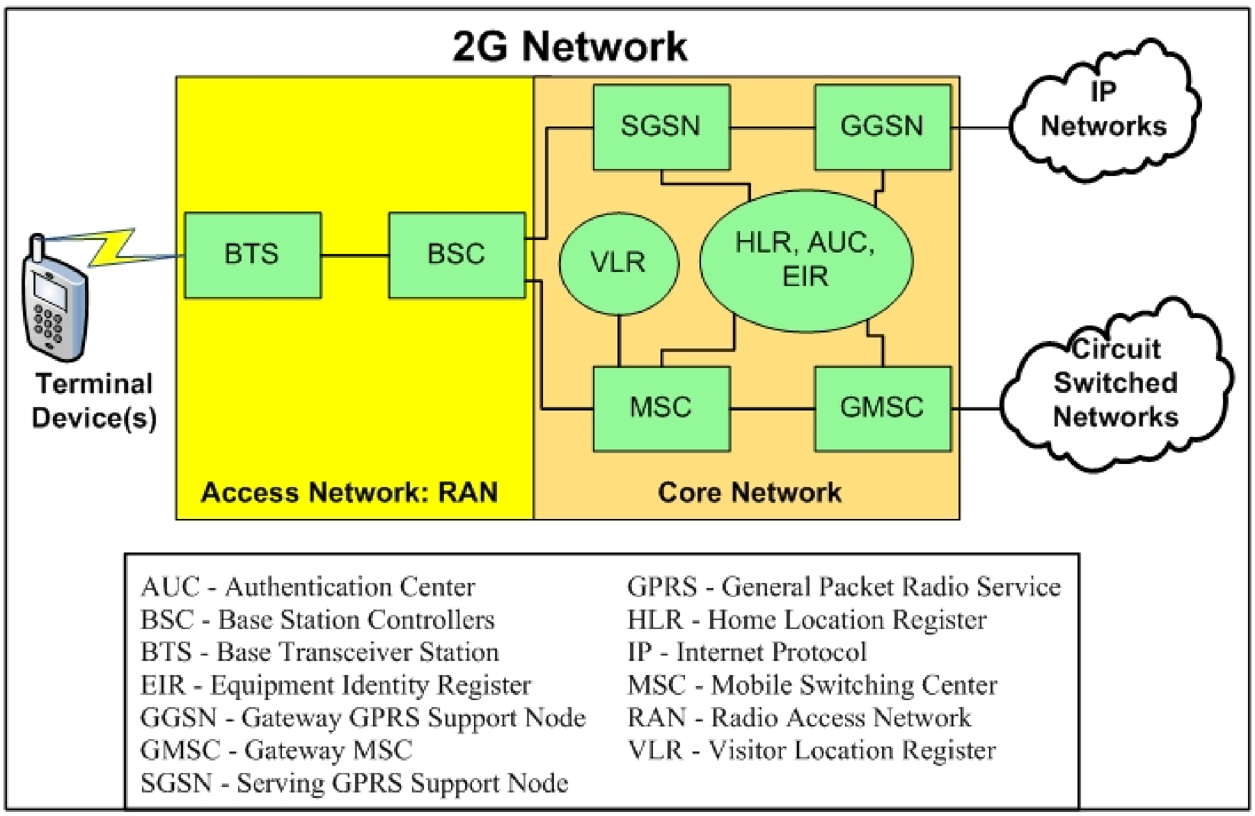

Figure 6.1.6-1. 2G Cellular Primary Components

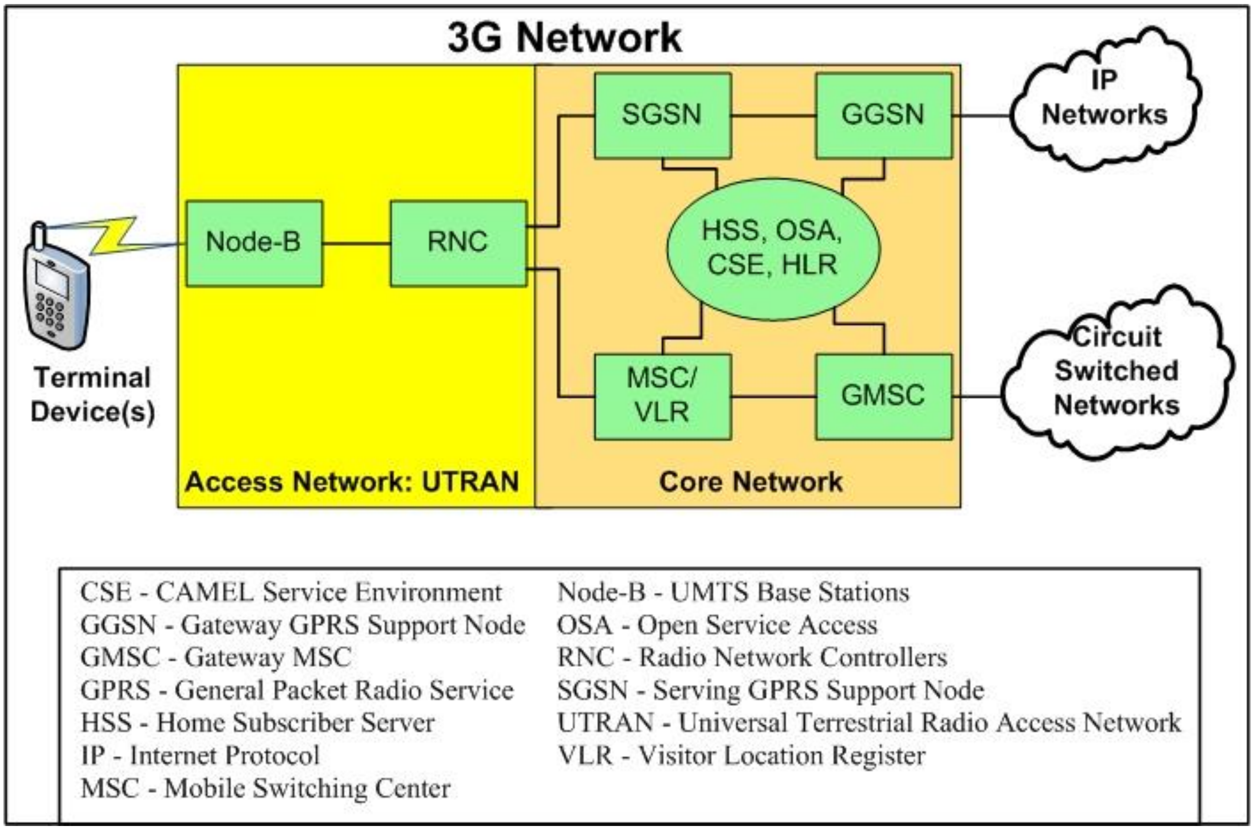

Figure 6.1.6-2. 3G Cellular Primary Components

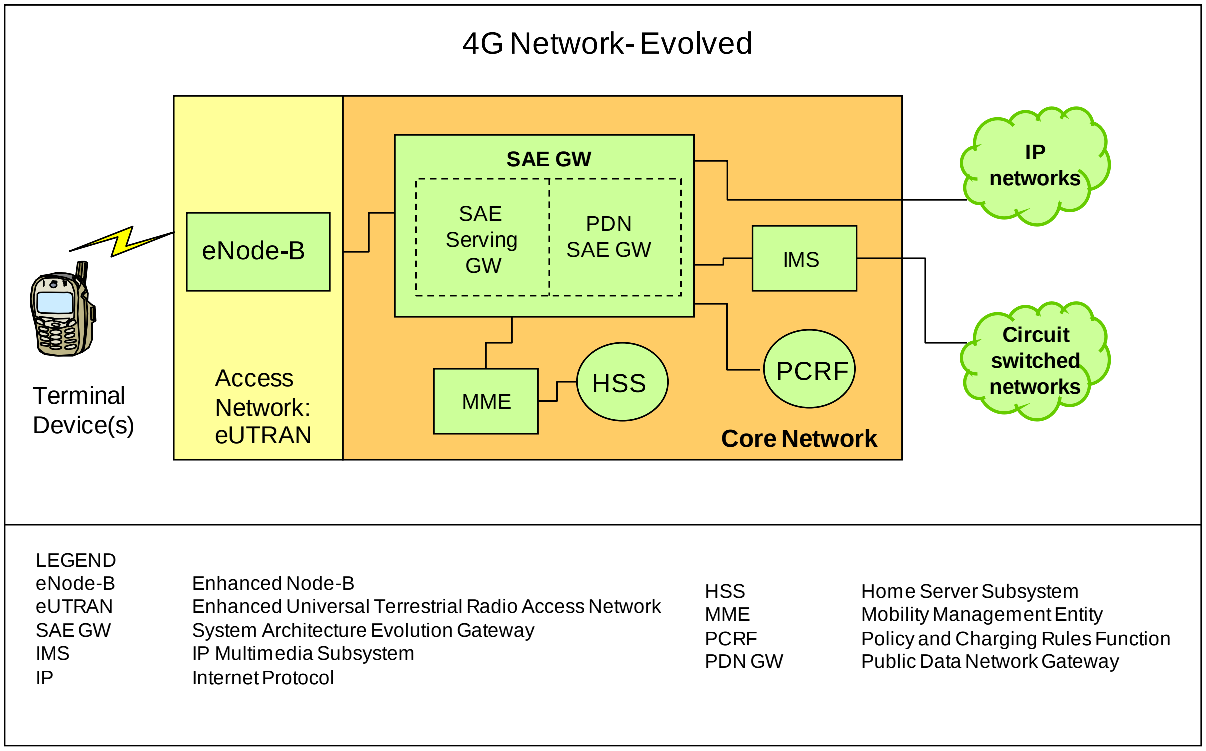

Figure 6.1.6-3. 4G Cellular Primary Components

6.1.6.5 DCVX Operation

The DCVX functions and provides mobile cellular services similar to standard commercial cellular systems with the addition of MUFs. It is based on a two-way cellular radio system that interconnects cell phones with other cell phones and landline stations. When used, the DCVX will provide full mobile cellular coverage in designated deployed environments; this includes training, exercise, and operational missions within COCOM AORs or specific geographic areas. User voice, data, and related communications via terminal devices will be similar to landline wired DSN or commercial services. Except for the inherent characteristics of radio transmission, basic service features between the two systems will be similar and transparent to the users. After full mature architectural implementation, the DCVX will function as a wireless adjunct and extension of the joint OAN tier of the GIG. The following configurations, illustrated in Figure 6.1.6-1, DCVX Connection Options, define the operational deployment options of a DCVX.

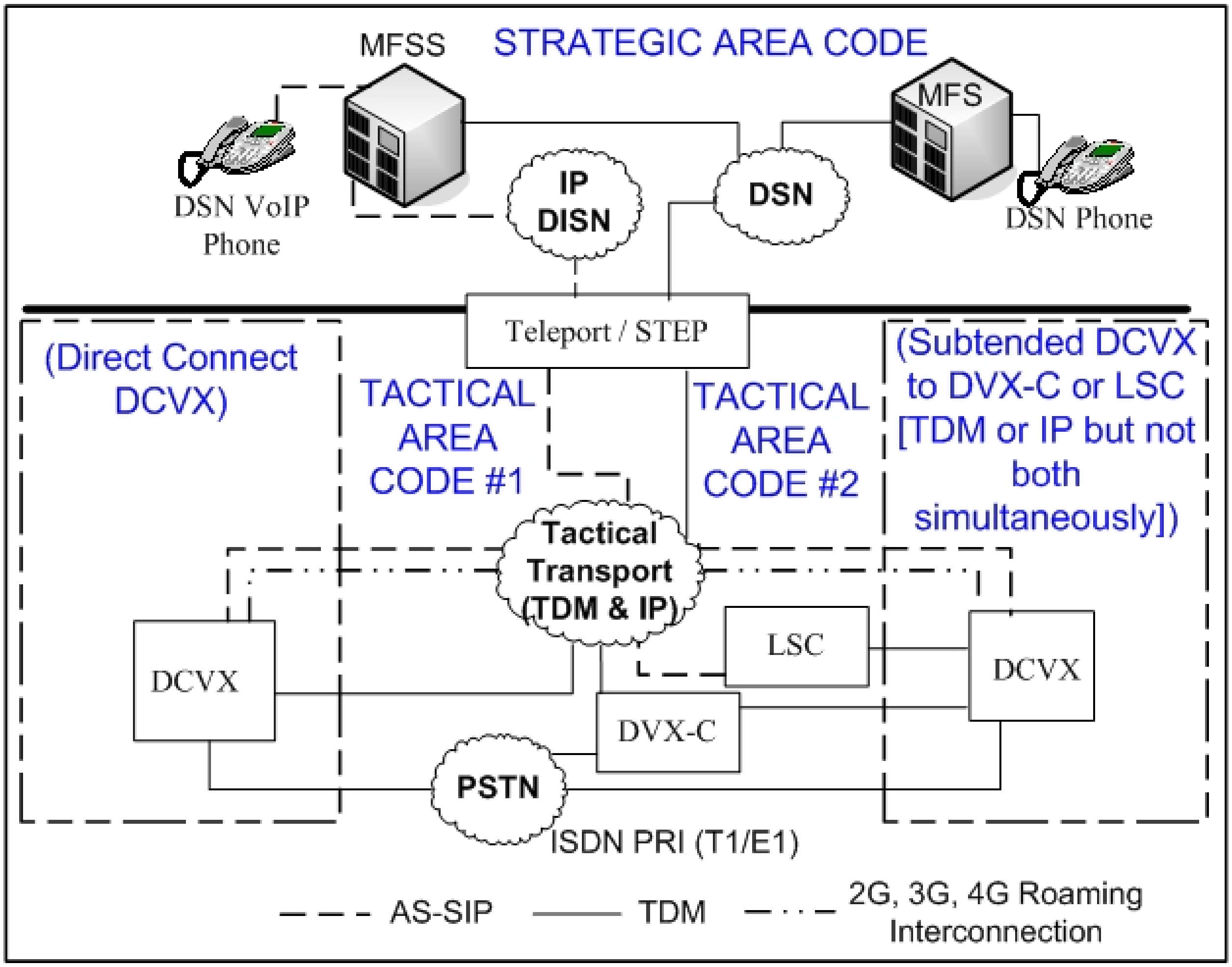

Figure 6.1.6-4. DCVX Connection Options

6.1.6.5.1 Subtended Deployment Connection

For a subtended deployed connection, the DCVX can reach DSN voice services or UC Services (VVoIP) using an existing authorized gateway switch, i.e., DVX-C or a Tactically deployed LSC, respectively. To accomplish this, the DCVX can connect to the Tactical TDM and IP transport networks, with one or more of the following interfaces:

- ISDN PRI (T1/E1)
- MLPP ISDN PRI (T1/E1)
- IP AS-SIP (TLS signaling and associated SRTP bearer channel)
- IP Non-UC Services (non-Real Time Data, i.e., Best Effort Data)

If the DCVX supports AS-SIP in this subtended configuration, connected to a Tactical LSC, then the DCVX operates in the Master-Subtended configuration to the Tactical LSC. The DCVX can support simultaneous interface connections to the DSN and UC VVoIP/Data networks using

TDM and IP respectively, but not use TDM and AS-SIP protocol simultaneously in support of voice and/or video calls. Current connections to the PSTN and/or other non-Government networks will be limited to ISDN PRI (T1/E1) only. Future IP-based PSTN voice and video service connections will be allowed once Information Assurance policy and STIGs are established.

6.1.6.5.2 Direct DSN Deployment Connection

For a direct DSN or UC VVoIP connections for UC Services, as well as IP data connections, the DCVX will use the "direct connection" configuration to the Tactical, TDM, and IP transport networks with one or more of the following interfaces:

- ISDN PRI (T1/E1)
- MLPP ISDN PRI (T1/E1)
- IP AS-SIP (TLS signaling and associated SRTP bearer channel)
- IP Non-UC Services (non-Real Time Data, i.e., Best Effort Data)

The DCVX can support simultaneous interface connections to the DSN and UC VVoIP/Data networks using TDM and IP respectively, but not use TDM and AS-SIP protocol simultaneously in support of voice and/or video calls. Current connections to the PSTN and/or other non-Government networks will be limited to ISDN PRI (T1/E1) only. Future IP-based PSTN voice and video service connections will be allowed once Information Assurance policy and STIGs are established.

6.1.6.5.3 Networked DCVX Deployment

When a DCVX is deployed in a networked DCVX configuration, a large deployed unit or multiple deployed units within the Tactical OAN may be interconnected with one or more HLR routing tables configured to support cellular terminal device roaming capabilities per the interconnections previously described.

For networked DCVXs within the Tactical OAN in support of a terminal device roaming capability, the DCVX configuration to the Deployed transport network will be with one or more of the following interfaces:

- ISDN PRI (T1/E1)
- MLPP ISDN PRI (T1/E1)
- IP AS-SIP (TLS signaling and associated SRTP bearer channel)
- SIGTRAN (CCS7 over IP)
- 2G, 3G, and/or 4G Standards interconnection protocols transported over DoD Networks

The extent of terminal device roaming capability will depend on the number and type of interconnections made between the DCVXs within the Tactical OAN and switch lookup routing table updates in the DCVXs themselves.

Current connections to the PSTN and/or other non-Government networks will be limited to ISDN PRI (T1/E1) only. Future IP-based PSTN voice and video service connections will be allowed once Information Assurance policy and STIGs are established.

6.1.6.5.4 Stand-Alone DCVX Deployment

When a DCVX is used in a stand-alone configuration, the only area served is a deployed unit establishing a JTF and its command, control, communications, and computers (C4) infrastructure. There is no DSN or PSTN access and no roaming beyond the deployed local network unit cell towers of its area of operation. The DCVX operates solely in an isolated mode.

6.1.6.6 General Description of Cellular Mobile Features and Technologies

6.1.6.6.1 Priority Access Service/Wireless Priority Service

Priority access service (PAS) provides the logical means for authorized mobile users to queue to the front and obtain priority access to the next available channel in a wireless call path. The goal of the wireless priority service (WPS) is to provide an E2E OAN-wide wireless priority communications capability to key military personnel during natural or manmade disasters. The WPS is an enhancement to basic cellular service. The full WPS capability can provide priority handling from mobile call origination, through the network, and all the way to the call destination.

The WPS is invoked by keying a special access number (*272) before the destination number on cellular instruments that have been classmarked for the WPS feature. A WPS user may be assigned one of five priority levels (i.e., 1, 2, 3, 4, or 5), with “1” being the highest priority level and “5” being the lowest. Each priority level has user-qualifying criteria that may be tracked for MLPP in DSN or in the UC Network via AS-SIP.

When a WPS call is queued for a radio traffic channel from a cellular user and no channel is available, the call is queued according to (1) the highest PAS priority first, and (2) queue entry time (i.e., earliest call first) within the same priority. If the queue for the call sector is full and the caller’s priority is determined to be higher than the level of the lowest priority caller in the queue, then the most recent WPS entry shall be removed, with the new WPS call request queued IAW items (1) and (2).

6.1.6.6.2 DoD Global System for Mobile Cellular Band

The current dedicated DoD Global System for Mobile (GSM) band is from 1755 MHz to 1835 MHz, which is a subset of the commercial DCS-1800 band. The remaining Government-owned frequency ranges are 1755 MHz to 1785 MHz for the uplink and 1805 MHz to 1850 MHz for the downlink. There are no non-DoD regulatory challenges associated with the use of the GSM band. The band has been approved for exclusive DoD use and is not authorized for use by any other entity. This band can be used for both voice and data applications to support unique DoD requirements. The Government-owned band may be adjusted in the future, and can be used appropriately at that time.

The band benefits are only effective in a CONUS environment; however, the DoD GSM may be used in OCONUS with specific host country/countries' authorization. The normal DoD frequency allocation process shall be followed to allow system operation within this band, and CC/S/A planners must ensure that an alternative solution is available before deployment as part of the planning process.

6.1.6.6.3 Precedence and Preemption

Precedence and preemption can only be implemented in a DoD network. This service has two parts: precedence and preemption. Precedence involves assigning a priority level to a call (wireless or wired). Preemption involves the seizing of a communications channel that is in use by a lower precedence level caller, in the absence of an idle channel. In the DCVX, the Precedence and Preemption capability is Conditional. Precedence and preemption may be provided by enacting enhanced multilevel precedence and preemption (eMLPP) or a vendor proprietary version that performs precedence and preemption in the DCVX between the terminal device and the cellular switch. The eMLPP is a cellular version of MLPP and Assured Service in TDM and IP networks respectively. In either version, precedence will be invoked by keying defined digits before dialing the destination number on cellular instruments that have been classmarked for this service. Precedence will function jointly in combination with WPS and will perform E2E as an adjunct to regular MLPP service on the wired DSN and Assured Service on the UC Network. However, in either of the provided versions, if available in the DCVX, eMLPP or vendor proprietary, the connection to the DSN will be MLPP PRI (T1.619a) or use the AS-SIP protocol for the UC Network.

Mobile systems, as currently designed, provide a maximum of seven priority levels. The two highest levels (A and B) are reserved for network internal use (e.g., for emergency calls or the network-related service configurations for specific voice broadcast or voice group call services). The second highest level (B) can be used for network internal use or optionally, depending on regional requirements, for subscription. These two levels (A and B) can only be used locally, that is, in the domain of one DCVX. The other five priority levels are offered for subscription and can be applied globally if supported by all related switch elements, and for interworking with

ISDN networks providing the MLPP service or Assured Service on UC Network. The seven eMLPP priority levels and their respective mapping to MLPP are defined as follows:

- **A** Highest, for network internal use
- **B** For network internal use or, optionally, for subscription
- **0** For subscription: FLASH-OVERRIDE
- **1** For subscription: FLASH
- **2** For subscription: IMMEDIATE
- **3** For subscription: PRIORITY
- **4** Lowest, for subscription: ROUTINE

Levels A and B shall be mapped to level "0" for priority treatment outside of the DCVX area in which they are applied. The vendor-proprietary version will support the five precedence levels as specified for DSN MLPP or UC Assured Service.

6.1.6.6.4 Code Division Multiple Access Mobile Systems

Mobile Code Division Multiple Access (CDMA) technology uses spread-spectrum telecommunications techniques in which a signal is transmitted in a bandwidth considerably greater than the frequency content of the original information. The latest technology today is based on third generation (3G) that allows high and fast bandwidth, generically called Evolution-Data Optimized (EVDO or EV-DO). This capability supports data usage of the terminal device to allow data connections to DoD networks and future possible use of a VoIP softphone on terminal devices when connected to commercial networks for extension of DSN single number presence.

6.1.6.6.5 GSM Communications Mobile Systems

Early technology for GSM allowed for the use Time Division Multiple Access (TDMA) technology. The TDMA allows several users to share the same frequency. It is the most popular standard for mobile phones in the world. The ubiquity of the GSM standard makes international roaming very common with "roaming agreements" between mobile phone operators. The latest GSM standard is based on an open standard that is developed by the Third Generation Partnership Project (3GPP).

6.1.6.6.6 4G IMT-Advanced System

Fourth generation (4G) refers to the fourth generation of cellular wireless standards. It is a successor to the 2G and 3G families of standards. The nomenclature of the generations generally refers to a change in the fundamental nature of the service, non-backwards-compatible transmission technology, and new frequency bands. The term 4G refers to an all-IP packet-switched network, mobile ultra-broadband (gigabit speed) access, and multi-carrier transmission. 4G is based on the ITU-R standard IMT-Advanced (*International Mobile Telecommunications Advanced*). An IMT-Advanced cellular system must have target peak data rates of up to approximately 100 Mbit/s for high mobility such as mobile access and up to approximately 1 Gbit/s for low mobility such as nomadic/local wireless access, according to the ITU requirements. The 3GPP and Worldwide Interoperability for Microwave Access (WiMAX) standards that will meet the ITU IMT-Advanced standard, are the pending 4G-Advanced and 802.16m respectively. In all suggestions for 4G, the CDMA spread spectrum radio technology used in 3G systems and IS-95, is abandoned and replaced by frequency-domain equalization schemes, for example multi-carrier transmission such as OFDMA. This is combined with Multiple In Multiple Out (MIMO) (i.e., multiple antennas), dynamic channel allocation and channel-dependent scheduling. In the meantime, pre-4G technologies such as first-release 4G Long Term Evolution (LTE) and Mobile WiMAX, have been available on the market since 2009 and 2006 respectively. However, 4G-LTE does not address the use of voice (a.k.a. VoIP) at this time. The GSM Association, via the Voice over LTE (VoLTE) initiative, is addressing this omission by selecting a subset of IP Multimedia Subsystem (IMS) standards to deliver E2E voice and SMS for LTE devices, including defining roaming and interconnect interfaces. In the meantime, most commercial cellular providers utilize Circuit-Switched Fallback (CSFB), which uses some initial signaling over the LTE Radio Access Network (RAN) and then "falls back" to the 2G/3G TDM RAN to establish the calls.

6.1.6.6.7 Secure Communications Interoperability Protocol

The SCIP is the NSA-approved secure voice and data encryption protocol used by DoD, U.S. Government agencies, and civilian authorities. The SCIP is used by NATO and coalition partners to provide secure voice interoperability between the United States and authorized foreign entities. Application of SCIP is described in detail in Section 5.2.2, DoD Secure Communications Devices.

6.1.6.7 DCVX Requirements Terminology

Requirements terminology is defined in Section 5.1.4, General Requirement Language.

6.1.6.8 DCVX General Requirements

6.1.6.8.1 Coverage and Signaling Strength

[Required] The signal strength shall not be less than the current GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and 4G authorized international standards and specifications. The GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and 4G technology are spectrum based; therefore, GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and 4G band, coverage, signal strength, and power are the basis for a planned "area of support." Environment, weather, geography, topography, and adjacent spectrums are elements that must be considered when applying the basis for an area of support. For testing purposes, the generic set of parameters presented in Table 6.1.6-1, Current Cellular Systems Parameters, shall be used for JITC certification either by testing and/or as determined by JITC.

Table 6.1.6-1. Current Cellular Systems Parameters

DCVX GSM/GPRS (2G, 3G, PRE-4G)	
Bands	As provided by standards and/or DoD GSM Cellular Band (e.g., 450 MHz, 850MHz, 900MHz, and 1900 MHz)
Specification on Coverage	As provided by standards (e.g., ITU-R 2G. 2.5G, 3G, 3GSM, UMTS, GSM Edge) (www.itu.int/publications)
Distance Transmit/ Receive	Up to 25 miles depending on topology/manmade structures, and frequencies also determine coverage parameters.
DCVX CDMA	
Bands	As provided by standards (e.g., 450 MHz, 700 MHz, 800 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, and 2100 MHz)
Specification on coverage	As provided by standards (e.g., TIA, IS-95, 3GPP2, IMT-2000, CDMA 1XRTT, CDMA2000) (www.tiaonline.org)
Distance Transmit/ Receive	Up to 32 miles depending on topology/manmade structures and frequencies also determine coverage parameters.
DCVX (4G IMT-Advanced)	
Bands	As provided by standards (e.g., GSM: 700 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, 2100 MHz and 2600 MHz, Mobile WiMAX: 500 MHz to 3.5 GHz)
Specification on Coverage	As provided by standards (e.g., GSM: 4G-Advanced, Mobile WiMAX, 802.16m, Pre-4G: 802.16-2009)
Distance Transmit/ Receive	Up to 25 and 30 miles for 4G-Advanced and WiMAX respectively depending on topology/manmade structures and frequencies also determine coverage parameters.
TERMINAL DEVICE	
Bands	As provided by standards (CDMA/GSM/4G-Advanced) and/or DoD GSM Cellular (e.g., 450 MHz, 700 MHz, 800 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, 2100 MHz, and 2600 MHz, Mobile WiMAX, 500 MHz to 3.5 GHz)
CDMA Specification	As provided by standards (e.g., CDMA (IS95), CDMA2000, CDMA 1XRTT and CDMA 1xEVDO)
GSM Specification	As provided by standards (e.g., GSM (GSM 02.07 Tech. Spec.(ver.7.1.0 Rel. 1998), 2.5G, 3G, 3GSM, GSM Edge)
4G Specifications	As provided by standards (e.g., GSM: 4G-Advanced; Mobile WiMAX; 802.16m, Pre-4G: 802.16-2009)
Distance Transmit/ Receive	Up to 8 miles depending on topology/manmade structures and frequencies also determine coverage parameters.

Table 6.1.6-1. Current Cellular Systems Parameters (continued)

LEGEND					
1xEVDO	One Times EVDO	CDMA2000	Code Division Multiple Access 2000	IMT-2000	International Mobile Telecommunications 2000
1XRTT	One Times Radio Transmission Technology	DCVX	Deployed Cellular Voice Exchange	IS-95	Interim Standard 95
3G	Third Generation	DMSC	Deployed Mobile Switching Center	ITU-R	International Telecommunication Union – Radiocommunication Sector
3GPP2	Third Generation Partnership Project 2	DoD	Department of Defense	MHz	Megahertz
3GSM	Third Global System for Mobile	EVDO	Evolution-Data Optimized	TIA	Telecommunication Industry Association
4G	Fourth Generation	GPRS	General Packet Radio Service	WCDMA	Wideband CDMA
BSS	Base Station Subsystem	GSM	Global System for Mobile	WiMAX	Worldwide Interoperability for Microwave Access
CDMA	Code Division Multiple Access				

6.1.6.8.2 Protocol/Format

[Required] The DCVX shall support at least one or more of the following protocols:

- GSM/ General Packet Radio Service [GPRS) (2G, 2.5G, Third Generation 3G), Third Global System for Mobile (3GSM), GSM Edge]
- Wideband CDMA (WCDMA)
- CDMA2000
- CDMA One Times Radio Transmission Technology (1XRTT)
- Universal Mobile Telecommunications System (UMTS)
- EVDO (or EV-DO)
- Mobile WiMAX [802.16-2009]
- Fourth Generation International Mobile Telecommunications-Advanced (IMT-Advanced)
- 4G-Advanced
- Mobile WiMAX Series [802.16m and beyond]

6.1.6.8.3 MOS and Measuring Methodology

[Required] The DCVX shall support the minimum MOS scores as defined in Section 5.3.3, Network Infrastructure End-to-End Performance Requirements, or better as measured in any 5-minute interval using ITU-T Recommendation P.862 testing standard. The baseline test environment shall be operated in an open air, clear of obstruction, line-of-sight environment, with the specific requirements as outlined in Table 6.1.6-1. Based on the results, the estimated MOS performance range will be extrapolated and provided in the vendor Letter of Compliance (LOC) based on the Access Network operating at or near full power mode and, at a minimum, operating at a height of 80 feet. The values provided in the vendor LOC will be included in the APL report. Refer to Section 6.1.6.14, Submission of Wireless Systems to UCAO for DSN Connection Approval Request, concerning guidelines on submitting the cellular engineering analysis package.

6.1.6.8.4 Availability

[Required] The DCVX shall have an availability of 99.97 percent, which includes scheduled maintenance.

6.1.6.8.5 Encryption

1. **[Conditional]** Depending upon which of the following encryption types a terminal device provides to support secure calls: SCIP, other NSA-accredited encryption scheme(s), and/or other required accredited encryption schemes as defined in appropriate cellular STIGs, the DCVX must provide appropriate radio and network transport bandwidth to support the terminal device encryption requirements contained in Section 6.1.6.9.4, Terminal Device Encryption.

2. **[Conditional]** If a secure call capability is provided in the terminal device(s), the DCVX shall support SCIP, other NSA-accredited encryption scheme(s), and/or required accredited encryption schemes as defined in the appropriate cellular STIGs. The DCVX that supports SCIP (also known as terminal device) will be required to go secure E2E with another SCIP Phone and/or via a SCIP Gateway if AS-SIP is used while the DCVX supports the establishment and maintenance of the secure call.

3. **[Conditional]** The DCVX may have the capability to provide secure SCIP Gateway functions.

6.1.6.8.6 Calling Features

6.1.6.8.6.1 Call Waiting Feature Requirement

The Call Waiting (CW) feature interacts with MLPP and Assured Service for TDM and IP, respectively. If a precedence and preemption capability is available in the DCVX, the preemption interactions must meet the requirements described in Section 6.1.6.8.10.1, Precedence Call Waiting. Call Waiting is a feature where a line in the talking state is alerted by a CW tone when another call is attempting to complete to that line. A CW tone is only audible to the line with the CW feature activated.

1. **[Required]** The CW feature shall generate a CW tone only audible to the line with the CW feature activated.

2. **[Required]** The Cancel CW feature is required when CW is active. The user must be able to cancel the CW service. Cancel CW is a feature that allows the user with CW service to inhibit the operation of CW for one call. The user dials the Cancel CW code, obtains recall dial tone, and places a call normally. During this call, the CW service shall be inactive so that anyone calling the CW user shall receive the normal busy treatment, and no CW tones shall interrupt the user's call.

6.1.6.8.6.2 Three-Way Calling Requirement

The Three-Way Calling (TWC) feature interacts with MLPP and Assured Service for TDM and IP, respectively. If a precedence and preemption capability is provided in the DCVX, the MLPP interactions must meet the requirements described in Section 6.1.6.8.10.2, Precedence TWC.

1. **[Conditional]** The TWC feature allows a station in the talking state to add a third party to the call without operator assistance. To add a third party to the call, the TWC customer places the other party on hold, receives recall dial tone, dials the third party's telephone number, and then takes the first line off hold to establish the TWC connection. This may occur at any time after the completion of dialing the second number joining the TWC. After the TWC connection has been established, the customer with the service activated may disconnect the last party added. The customer with the service activated may terminate the TWC call by disconnecting. If either of the other two parties hangs up while the service-activating customer remains off-hook, the TWC is returned to a two-party connection between the remaining parties.

2. **[Conditional]** The terminal device may support signaling to allow TWC.

6.1.6.8.6.3 Conference Calling

The Conference Calling feature is Conditional because it interacts with MLPP and Assured Service for TDM and IP, respectively. If precedence and preemption and conference calling capabilities are provided in the DCVX, the preemption interactions must meet the requirements described in Section 6.1.6.8.10.3, Precedence Conference Calling.

1. **[Conditional]** The Conference Calling feature allows the user to establish a conference call involving up to six conferees (including the user). This feature is requested via an access code.

2. **[Conditional]** The terminal device may support signaling to allow conference calling.

6.1.6.8.7 Roaming

[Conditional] The DCVX system may only support roaming to one or more DCVXs within the Tactical OAN per Section 6.1.6.5.3, Networked DCVX Deployment. The DCVX roaming numbering capability shall support the following:

1. Tactical Global Block Numbering Plan (GBNP)

2. Tactical Routing and Numbering: The DCVX shall be equipped and operationally capable of the dialing format for User Dialing Format to Coalition Forces as defined in Standard NATO Agreement (STANAG 4214), "International Rating and Directory for Tactical Communications Systems," Edition 3, Version T, 07 January 2005, or current edition.

Direct network connections from the DCVX to commercial cellular provider systems in support of terminal device roaming on the commercial cellular provider network(s) are not allowed.

6.1.6.8.8 Precedence and Preemption

The DCVX may support preemption and precedence under the following conditions:

1. **[Conditional]** The DCVX may support the cellular version of precedence and preemption, called eMLPP, and/or a proprietary methodology. When precedence and preemption are available, the interface to the DSN/UC Networks and/or the supporting DVX-C shall support one or more of the interfaces as described in Section 6.1.6.11.5.1, DCVX Core Network Trunks and Interfaces.

2. **[Conditional]** The DCVX will support a preemption and precedence capability under one or more of the following conditions:

    a. The DCVX supports GSM in the DoD GSM cellular band as described in Section 6.1.6.6.2, DoD Global System for Mobile Cellular Band.

    b. The DCVX supports the use of leased cellular frequency in one of the bands and protocol(s) listed in Table 6.1.6-1, Current Cellular Systems Parameters.

    c. The DCVX supports one or more of the cellular bands and protocol(s), as described in Table 6.1.6-1, Current Cellular Systems Parameters, in an OCONUS environment, where the local Forces-Status Agreement allows eMLPP/proprietary version operation.

d. The DCVX supports one or more of the cellular bands and protocol(s), as described in Table 6.1.6-1, Current Cellular Systems Parameters, dependent on the operational environment and usage of cellular frequencies allowed by local and/or US National Civilian Authorities.

6.1.6.8.9 Precedence Capability Terminal Device Activation/Deactivation

[Conditional] If a precedence and preemption capability is provided in the DCVX, the DCVX may be capable of providing on any supported terminal device the user's Precedence Class Table Assigned features. These features are provided to the terminal device based on the user entering a specified PIN on the same terminal device. The DCVX will assign to the terminal device the entire user's precedence capability as defined in the DCVX's class features table(s). This will allow the user to make precedence calls from terminal devices other than the one assigned or provided to the user. Additionally, the precedence features assigned to that active terminal device can be turned off by reentering the same or different PIN on the terminal device. The precedence capability user's activation or deactivation PIN may be stored in the DCVX or in another database accessible by the DCVX to validate the user's PIN(s) associated with the user's precedence capability. The user's precedence activation or deactivation PIN may be assigned and/or user settable after an initial assigned PIN has been provided.

6.1.6.8.10 Precedence and Preemption Calling Features

[Conditional] If a precedence and preemption capability is provided in the DCVX, then the following applies under the following calling features:

1. If no active call is in progress, the terminal device will receive precedence notification per Table 5.3.2.6-1, UC Ringing Tones and Cadences.

2. If a ROUTINE or lower precedence call is in progress to the terminal device, and a calling party calls at a higher precedence level, the current call will be preempted.

If a precedence call has been connected to the terminal device and is in progress, the calling party of equal or lower precedence will receive a notification that the lower precedence call was rejected. The following provides the precedence interactions for calls in progress to terminal devices.

6.1.6.8.10.1 Precedence Call Waiting

[Conditional] The following Precedence CW treatments shall apply to precedence levels of PRIORITY and above if the precedence and preemption capability is provided in the DCVX.

6.1.6.8.10.1.1 Busy with Higher Precedence Call

[Required] If the precedence level of the incoming call is lower than the existing precedence call, precedence CW shall be invoked. In an active call, if the incoming call is PRIORITY precedence or above, the precedence CW tone shall be applied to the called party per Table 5.3.4.10-4, UC Information Signals.

6.1.6.8.10.1.2 Busy with Equal Precedence Call

[Required] The DCVX shall provide the precedence CW signal to the called station per Table 5.3.4.10-4, UC Information Signals. The DCVX shall apply this signal regardless of other programmed features, such as call forwarding on busy or caller ID. The called station shall be able to place the current active call on hold, or disconnect the current active call and answer the incoming call.

6.1.6.8.10.1.3 Busy with Lower Precedence Call

[Required] The DCVX shall preempt the active call. The active busy station shall receive continuous preemption tone until an on-hook signal is received and the other party shall receive preemption tone for a minimum of 3 seconds. After the current call is terminated and the terminal device is idle, the station to which the precedence call is directed shall be provided precedence notification ring per Table 5.3.2.6-1, UC Ringing Tones and Cadences, or comparable vibration cadence. The station shall be connected to the preempting call after going off-hook.

6.1.6.8.10.1.4 No Answer

[Required] If, after receiving the precedence CW signal, the busy called station does not answer the incoming DSN call within the maximum programmed time interval, the switch shall treat the call IAW Section 5.3.2.2.2.1.2.5, Precedence Call Diversion.

6.1.6.8.10.2 Precedence Three-Way Calling (TWC)

1. **[Conditional]** If precedence and preemption and TWC are provided in the DCVX, the following TWC requirements apply:

    a. **[Required]** In TWC, each call shall have its own precedence level. When a TWC is established, each connection shall maintain its assigned precedence level. Each connection of a call resulting from a split operation shall maintain the precedence level that it was assigned upon being added to the TWC.

b. **[Required]** The DCVX shall class mark the originator of the TWC at the highest precedence level of the two segments of the call. Incoming calls to lines participating in the TWC that have a higher precedence than the higher of the two segments shall preempt unless the call is marked non-preemptable.

c. **[Required]** When a higher precedence call is placed to any one of the TWC participants, that participant receives the preemption tone per Table 5.3.4.10-4, UC Information Signals. The other two parties shall receive a conference disconnect tone. This tone indicates to the other parties that one of the other TWC participants is being preempted.

d. **[Required]** In a TWC call where each connection is established at a different precedence level, the precedence level of the participant who initiated the TWC call shall be assigned the highest precedence of the two connections.

6.1.6.8.10.3 Precedence Conference Calling

1. **[Conditional]** If precedence and preemption and conference calling are provided in the DCVX, then the following precedence conference calling requirement is required:

    a. **[Required]** All addresses shall be processed at a precedence level equal to that precedence level dialed by the conference originator.

        (1) If all conference bridges are busy, ROUTINE precedence conference call attempts shall be connected to a “line busy” tone per Table 6.2.7-2, CVVoIP Information Signals, and call attempts at precedence levels above the ROUTINE precedence shall re-examine all conference bridges on a preemptive basis.

        (2) A conference bridge that is busy at the lowest level of precedence stored for all units shall be preempted for a higher precedence conference call.

        (3) When a conference bridge is preempted, a 2-second burst of preemption tone per Table 5.3.4.10-4, UC Information Signals, shall be provided to the conferees on the existing conference. The existing connections to the bridge shall be dropped, and the bridge shall send an on-hook signal automatically to the associated switch ports to permit the new connections to be established.

        (4) Where the requesting precedence level is equal to or lower than the existing conference, the connection shall be denied, and the caller shall be provided a Blocked Precedence Announcement (BPA) per Section 5.3.2.6.1.1.2, Announcements.

6.1.6.8.10.4 Voice Mail

The Voice Mail feature interacts with precedence and preemption. If precedence and preemption capability and voice mail are provided in the DCVX or voice mail added externally, the precedence and preemption interactions must meet the requirements described in Section 5.3.2.2.2.1.2, Precedence Call Diversion.

[Conditional] The DCVX may provide ROUTINE calls only voice mail capability for users. Additional features, such as message forwarding, may be provided in addition to a basic voice mail capability provided they do not interfere with precedence and preemption if the capability is provided in the switch.

6.1.6.8.10.4.1 Precedence and Preemption Interaction with Voice Mail

1. **[Conditional]** If precedence and preemption is provided in the DCVX and voice mail capability is provided internally to the DCVX or connected externally to the DCVX as an adjunct, the following requirement applies:

   a. **[Required]** The DCVX shall divert all precedence calls above ROUTINE that are destined for voice mail IAW Section 5.3.2.2.2.1.2, Precedence Call Diversion.

6.1.6.8.11 Management Capabilities for Terminal Devices

[Required] The DCVX shall have the capability to manage its supported terminal devices as published in its users' database (e.g., HLR or MME) so it can assign, transfer, or terminate services, features, and calling capability to include telephone numbers for its terminal devices.

6.1.6.8.12 Security

[Required] All components of the DCVX shall meet security requirements as outlined in DoDI 8510.01 and the applicable STIG(s).

6.1.6.9 Terminal Device-Specific Requirements

Cellular handsets, often referred to as mobile subscribers, handsets, PDAs, Smartphones, BlackBerrys, and any other end user cellular devices, commercial- or Government- developed, are herein referred to as terminal devices. The terminal device is the interface between the user and the cell network. The terminal device can be a handheld unit, a mounted mobile device, or a fixed location device.

6.1.6.9.1 Terminal Device Requirements

1. **[Required]** The terminal device shall provide the following status information to the network:

   a. Powered on
   b. Moved to a new location
   c. Alerting
   d. Dialing

2. **[Required]** The terminal device shall display the following status information to the end user:

   a. Signal strength
   b. Battery capacity
   c. Roaming status
   d. Service not available
   e. Call progress status

3. **[Required]** If no STIG exists for the terminal device, the terminal device shall have the ability to provide key-locking ability to lock the terminal device's keypad and unlock the keypad after providing the appropriate key sequence or PIN entries as provided by the vendor in the terminal device. The lock and unlock key sequence or PIN shall be set by the user. If the user PIN is unavailable or not supplied, an administrator method, which can be vendor proprietary, shall unlock the terminal device.

4. **[Conditional]** The terminal device may have the capability to support WPS on commercial networks and/or DoD networks where provided when not connected to and functioning on a DoD precedence and preemption network.

5. **[Conditional]** Removable and Exchangeable Subscriber Identity Module (SIM): If a SIM card is utilized, the SIM card in commercially available terminal devices shall be removable and exchangeable into other similar commercially available terminal devices that are compatible with the DCVX system (applicable to a GSM-based system). This excludes secure terminal devices and other terminal devices not readily commercially available.

6.1.6.9.2 Terminal Device Signaling

[Required] The terminal device shall provide information to allow the DCVX to identify the terminal device when the terminal device is powered up, successfully registered, and in active call status.

6.1.6.9.3 Terminal Device Frequency Band Support

A terminal device that supports more than one frequency band has a high connection and reliability capacity.

1. **[Conditional]** A terminal device may support multiple (e.g., five) frequency bands as specified in Table 6.1.6-1, Current Cellular Systems Parameters, for each protocol supported in Section 6.1.6.8.2.

2. **[Conditional]** The terminal device may also support roaming and interconnecting with commercial cellular networks when operating outside the transmission range of the home based DCVX and other supporting DCVXs interconnected in support of roaming within the Tactical OAN.

6.1.6.9.4 Terminal Device Encryption

1. **[Conditional]** If SCIP and/or other NSA-accredited encryption are implemented in the terminal device, the SCIP and/or other NSA-accredited encryption-capable terminal device shall have the capability to go secure to provide E2E encryption to another secure cellular-capable terminal device, and via the DCVX, to a non-cellular NSA encryption-capable device per the requirements specified in Section 5.2.2, DoD Secure Communications Devices (DSCD). The SCIP and/or other NSA-accredited encryption device shall provide E2E encryption within the DCVX, from DCVX to DCVX (roaming) and from DCVX to external networks such as DSN, UC Network and/or PSTN.

2. **[Conditional]** The terminal device may support other non-NSA encryption schemas, such as AES encryption as used by the Government Emergency Telecommunications Service (GETS) system.

6.1.6.9.5 Terminal Device Battery Requirements

1. **[Required]** The commercially available nonsecure terminal device that is readily available must have a battery that shall provide as a minimum 6 days standby time in total and 3 hours nonsecure talk time in total but not both requirements sequentially on the same battery charge. The NSA encryption secure terminal devices (e.g., PDA Secure Mobile Environment Portable Electronic Device (SME PED)) must provide their specified battery and secure or nonsecure talk time. All other terminal devices must provide their specified battery and nonsecure talk time and/or secure talk time, if applicable.

2. **[Required]** The terminal device shall have the capability, when the primary battery is removed or drained, to retain primary network and user settings on the device before

another primary battery is installed or recharged. This is required to ensure the terminal device is able to reconnect to the DCVX upon power-up.

6.1.6.9.6 Terminal Device Secure Call Handling

[Conditional] If the terminal device supports SCIP or other NSA-accredited encryption scheme(s), the terminal device and/or DCVX system will provide classified secure call handling features, as defined in Section 5.9.3.8, Secure Call Handling, if conversion is made from TDM to IP Network boundaries.

6.1.6.9.7 Terminal Device Display and Alerting Features

The terminal device shall have the following display and alerting features:

1. **[Required]** Power-On Status. When the terminal device is powered on, the display shall indicate:

    a. Signal strength
    b. Remaining battery capacity
    c. Active call status
    d. Registration results (either success or failure)

2. **[Required]** ROUTINE Call Alerting. The idle, registered terminal device shall provide or be provided with an auditory and/or visual display alert for incoming ROUTINE calls.

3. **[Conditional]** Precedence Call Alerting. The DCVX may be required to meet the eMLPP functionalities specified in Section 6.1.6.6.3, Precedence and Preemption. The eMLPP references or uses a proprietary methodology. If precedence and preemption capability is provided, upon receiving a precedence call, the idle, registered terminal device will provide or be provided with a precedence alert and/or tone notification. Whether using eMLPP or a proprietary version, the terminal device shall issue the same alerting tone(s) for precedence calls IAW eMLPP requirements. Upon notification, the user will have the capability to select or reject the call of higher precedence.

6.1.6.10 Access Network-Specific Requirements

Specific Access Network capability is as follows:

6.1.6.10.1 Signaling

[Required] The Access Network will determine which channel to use for call setup IAW the appropriate supported protocols listed in Section 6.1.6.8.2, Protocol/Format, as outlined in Table 6.1.6-1, Current Cellular Systems Parameters.

6.1.6.10.2 Strength

[Required] The Access Network will monitor the terminal device for signal strength and transfer the terminal device to the stronger cell when necessary IAW the appropriate supported protocols listed in Section 6.1.6.8.2, Protocol/Format.

6.1.6.10.3 Protocol/Format

[Required] The Access Network shall support one or more of the protocols listed in the DCVX general requirements, Section 6.1.6.8.2, Protocol/Format and as outlined in Table 6.1.6-1. Current Cellular Systems Parameters.

6.1.6.10.4 Coverage

[Required] The Access Network will assign the strongest cell to the terminal device per the standards. The coverage area this system will provide shall be IAW the GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX and/or 4G standards and specifications IAW Table 6.1.6-1, Current Cellular Systems Parameters, and Section 6.1.6.8.2, Protocol/Format. Actual coverage will depend on topology and/or manmade structures and frequencies.

6.1.6.10.5 Preemption

[Conditional] If precedence and preemption capability is provided in the DCVX, in the event of a preemption for reuse, the Access Network and/or Core Network must disable the old call and maintain the current channel assignment to the terminal device in order to allow the set up of the new call. In the event where there are no idle channels available and preemption for reuse does not occur, then when a precedence call is received, the DCVX will find the lowest precedence channel and preempt that channel to allow for the higher-level precedence call to be completed.

6.1.6.11 Core Network-Specific Requirements

Due to the differences between the various cellular generations (2G, 3G, Pre-4G, 4G), it is not feasible to identify specific component requirements. Thus, this section will refer to Core Network functionality instead. Additionally, the home location registry (HLR) functionality is not required to be a local component part of the Core Network, but it will be necessary for the

Core Network to access a home location register at some location to determine the attributes of its supported terminal device. Whether the home location registry functionality is local with the Core Network or it is remotely queried, the home location registry functionality is a component of the DCVX under test.

6.1.6.11.1 Visitor Location Register Functionality

[Required] The Core Network shall maintain a Visitor Location Register Functionality to allow service to any authorized active terminal device within its domain per Section 6.1.6.8.2, Protocol/Format. VLR functionality may be updated by the DCVX resident HRL functionality, a shared HLR functionality with another DCVX, and/or via roaming between DCVXs.

6.1.6.11.2 Home Location Register Functionality

1. **[Required]** The Core Network shall connect to an HLR functionality to determine the attributes of the terminal device currently being served by the DCVX. The HLR Functionality can be co-located with the Core Network or accessed remotely. Access to the remote HLR Functionality may be by one or more of the following connection types:

    a. ISDN PRI (T1/E1)
    b. MLPP ISDN PRI (T1/E1)
    c. IP AS-SIP (signaling and associated bearer channel)
    d. SIGTRAN (CCS7 over IP)
    e. 2G, 3G, and/or 4G Standards interconnection protocols transported across DoD Networks.

2. **[Required]** <u>HLR Storage</u>. The HLR Functionality must store and support information on each terminal device registered to the network that the HLR Functionality serves.

3. **[Required]** <u>HLR Change and Propagation</u>. The HLR Functionality must support changes to the terminal device information. Once the HLR receives the supported change information, the HLR, whether local or remote from the Core Network, has 3 minutes to propagate the change information to the VLR Functionality. If the DCVX supports roaming, the HLR change must also propagate to the querying VLRs.

4. **[Conditional]** <u>Intra-DCVX Queries</u>. If a roaming capability is supported in the DCVX, the HLR Functionality must support queries from other DCVXs using specified protocol methods for obtaining terminal device information [e.g., GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and/or 4G standards] based queries.

6.1.6.11.3 Equipment Identity Register Functionality

[Required] To validate terminal devices to prevent a compromised terminal device from connecting to the cellular switch and obtain services, an EIR Functionality must be provided and integrated to work in conjunction with the Terminal Device Authentication Center Functionality as stated in Section 6.1.6.11.4, Terminal Device Authentication Center Functionality, to prevent compromising the DCVX.

6.1.6.11.4 Terminal Device Authentication Center Functionality

1. **[Required]** To authenticate terminal devices as valid terminal devices associated with the DCVX, the cellular switch will use standard cellular techniques, industry best practices, and/or vendor proprietary processes integrated into the switch.

2. **[Conditional]** Terminal devices not assigned to the supporting DMSC HLR (e.g., roaming terminal devices) may be supported for authentication via the industry standard(s) and/or industry best practices for roaming authentication.

6.1.6.11.5 Core Network External Network Trunks and Interfaces

[Required] The Core Network shall support one or more of the following TDM and/or IP trunks and interfaces. The Core Network can support simultaneous interface connections to the DSN and UC VVoIP/Data networks using TDM and IP respectively, but not use TDM and AS-SIP protocol simultaneously in support of voice and/or video calls.

6.1.6.11.5.1 TDM Support

1. **[Conditional]** If TDM trunks are supported, then the following requirements apply as directed:

   a. **[Required]** The Core Network will support ISDN PRI (T1/E1) as defined in Section 5.3.2.31.2, National ISDN 1/2 Basic Access for trunks that connect to the DSN/PSTN without MLPP capability.

   b. **[Conditional]** If a precedence and preemption capability is provided in the DCVX, the Core Network will support MLPP PRI (ANSI T1.619a, ITU Q.955.3 and/or Q.735.3) per Section 5.3.2.31.3.7, ISDN MLPP PRI

   c. **[Conditional]** The Core Network may support a DS1 Interface (e.g. PCM-24, PCM-30) per Section 5.9.2.3.4, DS1 Interface Requirements.

6.1.6.11.5.2 AS-SIP IP Trunking Support

[Conditional] If AS-SIP IP trunks are supported, then the DCVX shall comply with the stated requirements of an LSC, and if required, act as a SIP B2BUA. The Core Network and terminal devices supporting UC VVoIP Services are required to meet the conditions as stated in Section 5.4.5.4.9, Smartphones, Mobile UC Applications, and Backend Support Systems.

6.1.6.11.5.3 DCVX Interconnection (Roaming)

Including the connections provided in Section 6.1.6.11.5.1, TDM Support, and Section, 6.1.6.11.5.2, AS-SIP IP Trunking Support, one or more of the following connections can be used for connecting DCVXs together on DoD networks within the Tactical OAN in support of roaming capability and/or querying the local or remote HLR Functionality. Neither connection type below shall connect to the PSTN and/or other non-Government networks.

1. **[Conditional]** SIGTRAN: The Core Network may support CCS7 over IP using SIGTRAN IAW IETF RFC 2719, and other associated supporting RFCs.

2. **[Conditional]** 2G, 3G, and/or 4G Standards: The interconnection portion of the protocols contained within the 2G, 3G, Pre-4G, Wideband WiMAX, and/or 4G Standards, as delineated in Section 6.1.6.8.2 Protocol/Format, may be used to interconnect DCVX systems when said protocols are transported over DoD operated and/or controlled networks.

6.1.6.11.6 Non-MLPP Networks Support

[Conditional] The Core Network may support an ISDN PRI (T1/E1) non-MLPP trunk for connecting to the PSTN and/or other non-Government networks. ISDN PRI (T1/E1) requirements are contained within Section 5.3.2.31.2, National ISDN 1/2 Basic Access.

6.1.6.11.7 Call Handling

[Required] The Core Network shall handle both intraswitch calls and calls to and from the DSN, PSTN, and/or UC Services Network, while recognizing a powered-on terminal device that comes into its operational area.

6.1.6.12 Security

[Required] All components of the DCVX shall meet security requirements as outlined in DoDI 8510.01 and the applicable STIG.

6.1.6.13 DCVX Network Management

[Required] The DCVX is to be managed by at least one or more of the following:

1. **[Conditional]** A front or back panel and/or external console control capability shall be provided for local management.

2. **[Conditional]** Remote monitoring and management by the Advanced DSN Integrated Management Support System (ADIMSS) or similar NM systems developed by DoD Components. The following requirements apply:

   a. **[Required] Data Interface**: The NE shall provide NM data/monitoring via one or more of the following physical interfaces:

      (1) Ethernet/TCP/IP (IEEE 802.3)
      (2) Serial (RS-232)/Asynchronous
      (3) Serial/Synchronous (X.25 and/or BX.25 variant)

      All data that is collected shall be accessible through these interfaces. For NM purposes, the NE must provide no less than two separate data channels. They may be physically separate (e.g., two distinct physical interface points) or logically separate (e.g., two user sessions through a single Ethernet interface). The data may be sent in ASCII, binary, or hexadecimal data or ASCII text designed for screen/printer display.

      The data channels shall be used for and, as such, must be capable of providing:

      (1) Alarm/Log Data
      (2) Accounting data (e.g., CDR)
      (3) Performance Data (e.g., traffic data)
      (4) DCVX access (to perform DCVX data fill administration and network controls)

   b. **[Required] Fault Management**: The DCVX shall detect fault (alarm) conditions and generate alarm notifications. The alarm messages must be sent to the assigned NM Alarm channel in near-real time. No alarm restriction/filtering is necessary. In addition to the data formats in Section 5.9.2.4, Device Management, alarms may be sent as Simple Network Management Protocol (SNMP) traps. If this channel is also used to output switch administrative log information, the alarm messages must be distinguishable from an administrative log message.

c. **[Required] Configuration Management**: Requirements for this feature shall be in accordance with Telcordia Technologies GR-472-CORE, Section 4.

6.1.6.14 Submission of Wireless Systems to UCCO for DSN Connection Request

[Required] The DCVX systems shall be engineered so that the Access and Core Networks achieve the required performance requirements in their specific deployed environment. The user shall submit a network design and engineering performance analysis with supporting calculations to meet minimum MOS performance with the request for DSN, PSTN, and/or UC Services Network connection. For certification procedures, the UCCO submittal shall include wireless security compliancy as identified in Section 6.1.6.12, Security.

6.1.7 Deployed Tactical Radio Requirements

The requirements discussed in this section refer to post-2012 system deployments. This section does not discuss transition between current system deployments and the systems described herein.

6.1.7.1 Introduction and Purpose

The following sections describe the requirements that shall be met by all deployed Tactical Radio Networks (TRNs) for them to be certified and used in the OAN tier of the GIG. Requirements are defined at the system level, as well as the various components that make up the radio networks, including protocol requirements. Several of these requirements reflect changes described elsewhere in this UCR. These will be indicated in the text.

The scope of this section is limited to push-to-talk (PTT) TRNs. Future updates will address radios that have the ability to dial directly to a DISN VoIP EI.

6.1.7.2 Applicability

The requirements within this section are applicable to all PTT-based TRNs that connect directly or indirectly to the DISN VoIP services.

The current version of the UCR is the governing requirements document that takes precedence over the explicit or implicit requirements of subsidiary or reference documents, standards, and specifications, except for those requirements specified in the documents listed in Section 6.1.6.3, Policy and Reference Documents.

6.1.7.3 Policy and Reference Documents

The policy and instruction documents in Section 6.1.6.3 will in conjunction with the UCR be used as a basis for APL certification.

6.1.7.4 TRN System Overview

The TRNs provide wireless communication services with MUFs. They differ from commercial, standards-based cellular networks in that individual radios within the TRN can communicate with each other, without the need for a base station, BSC, or constituent signaling, or interconnect equipment. The TRNs use multicast-based Radio Frequency (RF) transmissions to enable a set of radios using the same frequencies to communicate with each other.

The lower portion of Figure 6.1.7-1, TRN Connectivity, shows the architecture for a notional TRN and how the TRN connects to the DISN UC-compliant service. This connection is facilitated by a new UCR function called the Radio Bridge Function (RBF). The RBF is a component of a deployed LSC, or a component of a radio within the TRN.

The upper portion of Figure 6.1.7-1 shows elements of the UC-compliant network and interconnects, as described elsewhere in this UCR.

The TRN, also known as a "Voice Net" for this description, is composed of Voice Net Segments. Each Voice Net Segment is a group of radios, which communicate on a common set of frequencies. At least one radio in each Voice Net Segment is designated as a Voice Net Access Radio (VNAR).

[Conditional] A VNAR performs as many as three roles, depending on the type of Voice Net. It acts as a conventional radio to communicate with other radios in its Voice Net Segment. If there is more than one Voice Net Segment in a Voice Net, the VNAR shall communicate with VNARs in the other Voice Net Segments using, what could be, a proprietary, packet-based radio access network (RAN). At least one VNAR in a Voice Net also may act as a UCR-compliant (e.g., APL-listed) EI, to enable voice communications between the Voice Net and UCR-compliant VoIP EIs.

The methods and formats for VNAR communications over the RF links and the RAN depend on the type of technology used to create the Voice Net. Some Voice Nets operate in multicast PTT mode, where one party speaks and the others listen. Access to the radio links is controlled by Layer 2 access methods used by all radios in the Voice Net Segment, and by human protocols. Communications are half-duplex either by design or by enforcement of human protocol. Other Voice Nets support point-to-point communications initiated by one radio connecting to one or a few designated radios using full duplex communications. This version of the UCR is limited to describing requirements for the support of PTT-based voice networks.

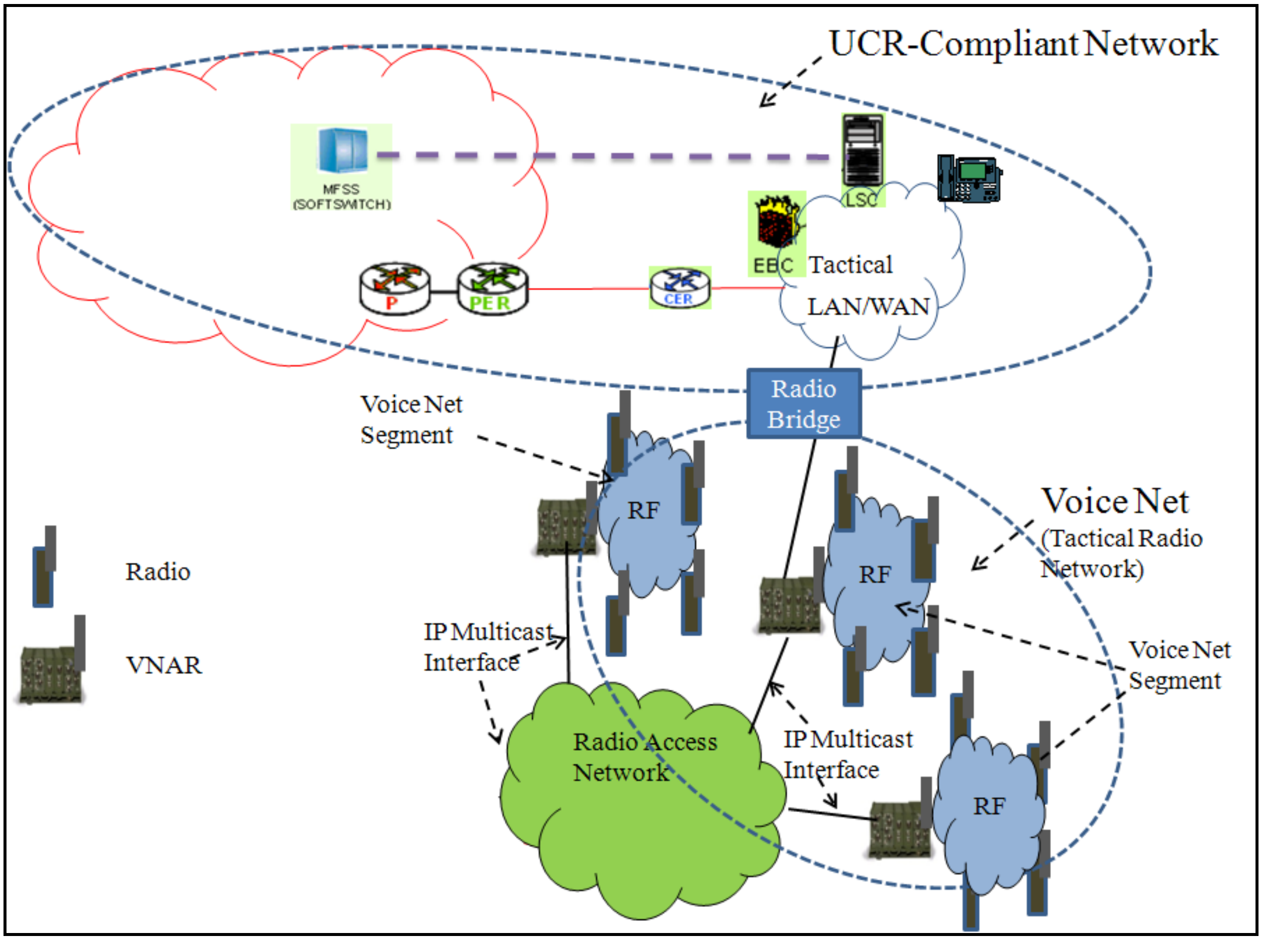

Figure 6.1.7-1. TRN Connectivity

Voice Net technology is not standardized. It is not the purpose of this UCR to create such standards. The requirements in this UCR are directed toward the communications between a VNAR and a UC-compliant voice EI. The basic requirement is that a VNAR provide a standardized EI interface so that other EIs can connect to the VNAR and become parties to the Voice Net. Figure 6.1.7-1, TRN Connectivity, shows the VNAR connected directly to an ASLAN. However, the VNAR also could connect via the RAN to a device that connects to an ASLAN.

6.1.7.5 Functional Description

This section defines the requirements for VNAR UCR compliance and requirements changes to enable Strategic and Deployed LSCs to support traffic flow between UCR-compliant EIs and VNARs.

Figure 6.1.7-2, Functional Connectivity, provides an overview of the major functions necessary to provide connectivity between VoIP EIs and a TRN. This figure does not include the specifics of how the functions connect to each other.

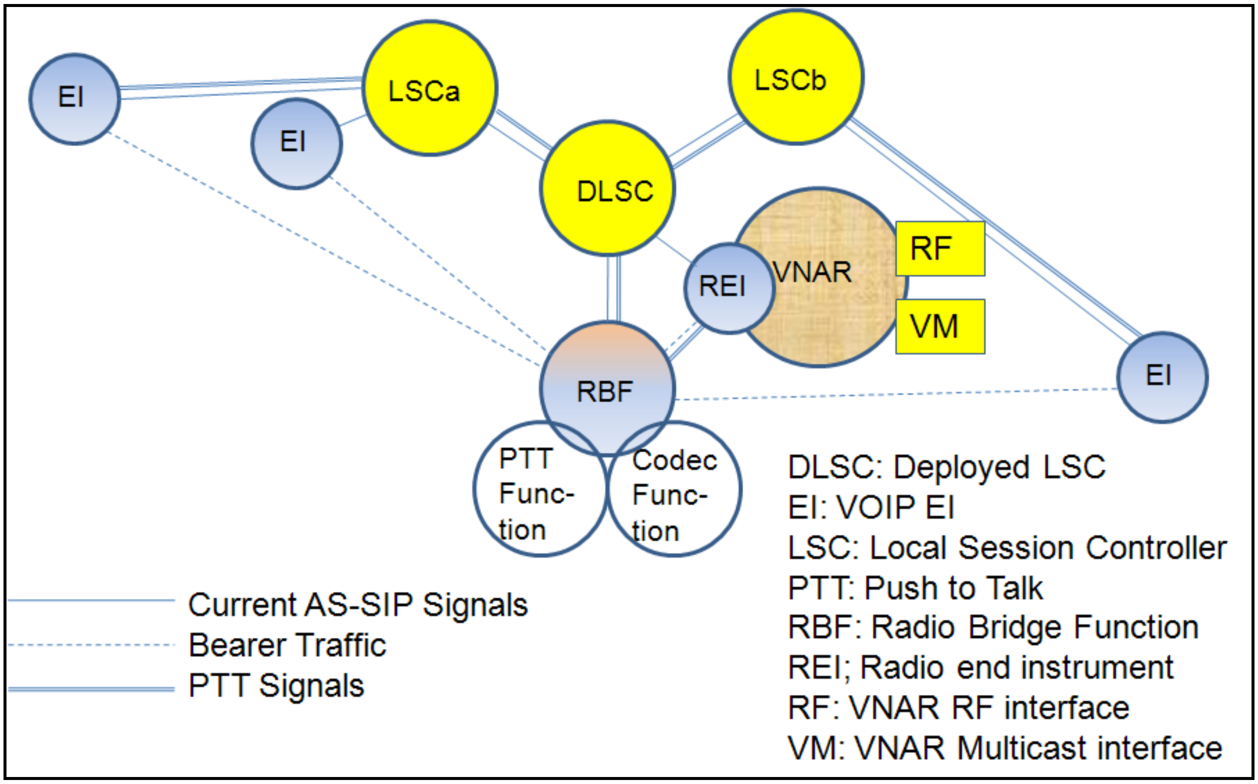

Figure 6.1.7-2. Functional Connectivity

The flow, shown in Figure 6.1.7-2, Functional Connectivity, assumes that PTT signaling will be sent out-of-band, via the LSCs using the concepts defined in RFC 4730. Alternatively, the PTT signaling could be sent in-band, using the concepts defined in RFC 4733. In such case, the tone signals will be sent along the bearer path between the EI (or a proxy for the EI) and the RBF.

The EIs and LSCs connect over a UCR-compliant IP network, as indicated in Figure 6.1.7-1, TRN Connectivity. The EIs and LSCs perform the functions defined for these elements as described within this UCR in Section 4.4.1.1.2.2, LSC Designs – Voice. In addition, the EIs and LSCs shall be enhanced to support a new set of AS-SIP signals that support PTT requirements for Tactical radios. The LSCa and LSCb in Figure 6.1.7-2 support fixed EIs. The deployed LSC (DLSC) in the figure supports deployed TRNs and cellular networks as described in Section 6.1.6, DCVX System Requirements, and for DLSC (TBD).

The RBF provides connectivity between the EIs and the Radio End Instrument (REI). It is an enhanced version of a conventional conference bridge described in Section 5.3.2.32, UC Audio and Video Conference Bridge Requirements.

The REI provides connectivity between the UCR-compliant domain and the Voice Net. The connection could require multiple router and switch hops. The RBF and REI also connect to the DISN IP voice network. The connection could be a LAN, WAN, or a back plane (if the functions are collocated in the same physical device).

The RBF could be a stand-alone appliance, or incorporated in a DLSC or incorporated in a VNAR. The PTT function, associated with the RBF, emulates the PTT function of the TRN to enable a conference participant to access the Voice Net. The codec function associated with the RBF performs transcoding as necessary, to enable media transfer between the REI and the EIs. The REI is always located within a VNAR. The VNAR is connected to an RF element, and possibly a RAN.

The EIs and REIs join a conference, which is dedicated to the Voice Net. The RBF provides half-duplex, PTT access for the EIs to the Voice Net and enables a conference participant to speak to the Voice Net. The AS-SIP is enhanced to provide a new set of signaling functions that support PTT access as described in Table 6.1.7-1, Control Information: VNAR to VoIP EI, and Table 6.1.7-2, Control Information: VoIP EI to VNAR.

Voice Net bearer traffic, shown as dotted lines in Figure 6.1.7-2, Functional Connectivity, flows between the REI and the RBF where it is replicated for transmission to each EI that has been admitted to the conference. Bearer traffic flows from each EI to the RBF. In general, this traffic is blocked at the RBF. The only exception is bearer traffic from an EI that has been granted access via the PTT function.

Current AS-SIP signaling traffic is shown as solid lines in Figure 6.1.7-2, Functional Connectivity. The PTT signaling traffic is shown as double lines in Figure 6.1.7-2, Functional Connectivity. Both types of signal traffic flow between an EI and its Master LSC (MLSC), from LSC to DLSC (possibly via MFSSs, which are not shown in the figure) to the RBF and the REI and in the reverse direction. The PTT signaling traffic also flows in both directions between the RBF and the REI.

6.1.7.5.1 Radio Bridge Function

1. **[Required: RBF]** The system shall support three types of participants in a "Meet-Me Bridge" Voice Net conference (see Section 5.3.2.32, UC Audio and Video Conference Bridge Requirements):

    a. Conference participant – An individual who joins the Voice Net using VoIP EIs

    b. VNAR participant – One or more VNARs connected to the Voice Net

c. Conference manager – A conference participant who has the authority to manage certain features of the conference

Each participant joins the conference by calling in to a unique telephone number or URI that is assigned to the conference. The RBF will be assigned a unique IP address that corresponds to its telephone number and URI.

2. **[Required: RBF]** The system shall lead the caller through an authentication process using voice messages. If the process determines the caller is authorized to join the conference, the system shall send a voice message informing the caller that he or she is now a conference participant and indicate the status of the conference. Status shall include:

   a. Conference is not yet available.

   b. Conference has been terminated.

   c. Conference is in process including the number of participants, and an indication if the VNAR is a participant.

3. **[Required: RBF]** If the authentication process determines that the caller is unauthorized, the system shall send a voice message informing the caller that he or she is ineligible for the conference. If the caller does not hang up in a parameter-determined period, the system will terminate the call, whereas the parameter termination period shall be determined using a configurable time-out parameter with a time-out range of 0–60 seconds; default shall be set to 5 seconds.

4. **[Required: RBF]** The authentication process shall include a participant code, which identifies the type of participant, and a log-in process suitable to the security level of the conference.

5. **[Required: RBF]** The system shall perform the following functions in addition to those described in Section 5.3.2.32, UC Audio and Video Conference Bridge Requirements:

   a. In the default mode, a conference participant is placed in a listen-only mode, where the participant can only hear audio transmitted from the VNAR.

   b. The system performs whatever codec transformations are necessary to ensure compatible communications between the VNAR and the EIs (see Section 6.1.7.5.4, Codec Translation Functional).

   c. The system supports the PTT function defined in Section 6.1.7.5.3, Push-to-Talk Functional Requirements. The PTT function will ensure that only one conference participant can speak to the Voice Net at a time, and only when there is no other party speaking on the Voice Net (see Section 6.1.7.5.3).

d. The conference manager has the ability to block or preempt any participant from access to the Voice Net.

e. The conference manager has the ability to bridge the conference participants to each other, so they can speak with each other. The traffic resulting from this bridging will not be transmitted to the Voice Net.

f. The conference manager has the ability to speak to any or all conference participants.

g. The conference manager has the ability to terminate the speaker-listener status of any conference or VNAR participant.

6. **[Required: RBF]** The system shall operate as an AS-SIP EI for authenticating, registering, and interacting with the LSC to originate or terminate voice sessions.

   NOTE: The RBF exchanges AS-SIP signaling packets with its Master DLSC. The DLSC exchanges AS-SIP messages with other LSCs or MFSSs to complete and tear down calls. The Master DLSC could be collocated with or remote from the RBF. The Master DLSC could be assigned uniquely to the RBF, or could support multiple RBFs.

7. **[Required: RBF]** The system shall provide a method for multiple VoIP EI users to concurrently connect to the Voice Net, up to a configurable limit.

8. **[Required: RBF]** The system shall allow automatic termination of the session, based on configurable events, including inactivity on an AS-SIP session for a specified session time limit.

9. **[Required: RBF]** The system shall support MLPP requirements if more calls arrive than can be supported (see Section 5.3.2.31.3, Multilevel Precedence and Preemption). The system shall be able to preempt a call from a lower precedence conference participant, if necessary, to provide resources to accept a call from a higher precedence conference participant if that call would otherwise be blocked. The system shall not preempt a call from a VNAR participant unless directed to do so by the conference manager.

10. **[Required: RBF]** The system shall support a configurable number of simultaneous conference participants per Voice Net.

11. **[Required: RBF]** The system shall support a configurable number of simultaneous VNAR participants per Voice Net.

12. **[Required: RBF]** The conference manager shall select one VNAR to act as master.

13. **[Required: RBF]** When the system is not acting as a master, the system shall act as a backup, which is available to replace the master system if the master fails. It is highly desirable that the system be implemented to support an automatic failover to a backup VNAR if a master VNAR or its connection fails.

    NOTE: The number of VNAR and conference participants is unspecified. These numbers are left to best design and engineering practices as determined by the supplier of the RBF function to meet the performance and reliability goals required for a particular deployment.

6.1.7.5.2 Radio End Instrument

1. **[Required: REI]** The system which resides within the VNAR, shall act as a conventional UCR-compliant EI in performing the following features defined in the following sections:

    a. Point-to-Point Call (Section 5.3.4.13.2, Point-to-Point Call, and Section 5.3.4.19.2, Point-to-Point Call)

    b. Tracing of terminating call (see Section 5.3.2.2.2.2.2, Tracing of Terminating Calls)

    c. Outgoing call tracing (see Section 5.3.2.2.2.2.3, Outgoing Call Tracing)

    d. Tracing of a call in progress (see Section 5.3.2.2.2.2.4, Tracing of a Call in Progress)

2. **[Required: REI]** The system shall transform bearer traffic packets received from the RBF to the format associated with its Voice Net and transmit the traffic to the Voice Net.

3. **[Required: REI]** The system shall transform bearer traffic received from the Voice Net to bearer traffic packets for transmission to the RBF.

4. **[Required: REI]** The system shall transform UCR standard PTT signals defined in Section 6.1.7.5.3, Push-to-Talk Functional Requirements, to the PTT signals required by the VNAR to support the Voice Net.

5. **[Required: REI]** The system shall transform PTT signals received from the Voice Net to UCR standard PTT signals for transmission to the RBF.

6. **[Required: REI]** The system shall operate as an AS-SIP EI for the purpose of authenticating, registering, and interacting with its Master DLSC to originate or terminate voice sessions.

NOTE: The DLSC could be colocated with or remotely from the REI. The DLSC could be assigned uniquely to the REI, or it could support multiple Voice Nets. The DLSC provides an MLPP function to give priority to higher precedence callers, if there is insufficient capacity to support a new dial-in call.

7. **[Required: REI]** The system shall support incoming session setup requests from IP EIs according to the AS-SIP specification [Reference: "Department of Defense Assured Service Session Initiation Protocol (AS-SIP) Generic System Requirement (GSR)"].

8. **[Conditional: REI]** The system shall support a call to a VoIP EI directly without going through an external RFB. If this option is invoked the REI shall support:

    a. The PTT function described in Section 6.1.7.5.3, Push-to-Talk Functional Requirements.

    b. The codec translation function described in Section 6.1.7.5.4, Codec Translation Function.

    c. The MLPP functions described in Section 5.3.2.31.3, Multilevel Precedence and Preemption.

    d. Three-way calling as described in Section 5.3.2.2.2, Three-Way Calling.

6.1.7.5.3 Push-to-Talk Functional Requirements

The VNAR, RBF, LSCs, and REIs cooperate to provide a capability that will enable a VoIP end user to initiate and terminate the equivalent of a PTT session.

1. **[Required: RBF]** The system shall provide a fail-safe mechanism to prevent a VoIP EI from streaming continuous voice traffic to a PTT-based Voice Net.

2. **[Required: RBF]** The system fail-safe mechanism shall ensure that, no matter the status of the VoIP end user or the VoIP EI, transmissions from the VoIP EI will terminate within a configurable, parameter-driven amount of time

3. **[Required: RBF]** The system fail-safe mechanism shall only reinstate transmissions based on completion of a specific, positive action by the VoIP end user.

4. **[Required: VNAR, RBF, REI – Conditional: LSC or DLSC]** The cooperating elements in a PTT session shall support DTMF tones after a call has progressed to the media session mode.

5. **[Required: RBF]** If PTT is configured, the system shall prevent bearer traffic generated from a VoIP EI, from accessing the Voice Net until the VoIP end user initiates a PTT session by entering a unique configurable tone sequence called the "talk tone." This action mimics depression of the PTT button on a radio, thereby initiating an emulated PTT session. The VoIP end user enters a different, configurable tone sequence ("end tone") to end the PTT session, emulating the release of the PTT button.

6. **[Required: RBF]** Upon receipt of the Talk Tone, the system in cooperation with the REI, shall determine whether the Voice Net is busy or available. The Voice Net is busy if any other party has been granted authorization to speak on the Voice Net. This can happen in one of two ways: 1) another conference participant has access to the Voice Net, or 2) a radio user has access to the Voice Net.

The following examples present traffic flow for the two cases where the Voice Net is busy, and the case where the Voice Net is available. These examples are representative, but not exclusive. Traffic flows could vary based on the type of technology used in the Voice Net. The examples assume out-of-band signaling and the use of VNARs that can provide tone responses indicating a Voice Net available condition. The flow of traffic for the first busy case is as follows:

1. The participant keys in the talk tone sequence.

2. The talk tone is sent from the originating EI to its MLSC.

3. The LSC converts the EI-generated talk tone to the UCR standard talk tone packet.

4. The LSC sends the talk tone packet to the DLSC to which the RBF is registered.

5. The RBF's Master DLSC sends the talk tone packet to the RBF.

6. The RBF determines that another conference participant has access to the Voice Net and cannot be preempted.

7. The RBF generates a standard "busy tone" packet for transmission to its Master DLSC.

8. The DLSC sends the busy tone packet to the LSC to which the requesting EI is registered.

9. The EI's MLSC converts the tone to a form that is supported by the requesting EI.

10. The requesting EI creates an audio signal indicating that the Voice Net is busy.

The flow of traffic for the second busy case is as follows:

1. The participant keys in the talk tone sequence.

2. The talk tone is sent from the originating EI to its MLSC.

3. The LSC converts the EI-generated talk tone to the UCR standard talk tone packet.

4. The LSC sends the talk tone packet to the DLSC to which the RBF is registered.

5. The Master RBF sends the talk tone packet to the RBF.

   a. The RBF determines that no other conference participant has access to the Voice Net.

   b. The RBF sends the talk tone packet to the REI.

   c. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net and the Voice Net sends a signal back to the VNAR indicating that the Voice Net is busy.

   d. The REI, within the VNAR, generates a standard busy tone packet and sends it to the RBF.

   e. The RBF sets a flag indicating that the Voice Net is busy and sends the standard busy tone packet to its Master DLSC.

   f. The DLSC sends the busy tone packet to the EI's MLSC.

   g. The LSC converts the tone to a form that is supported by the requesting EI, and sends the tone to the EI.

   h. The requesting EI creates an audio signal indicating that the Voice Net is busy.

   NOTE: There is a timing issue in the busy cases. In some situations, the delay between the time of the initial VoIP EI PTT request and the time the request arrives at the VNAR could be in the 1–2 second range. This could lead to a situation where speakers on the TRN could block out VoIP speakers. To mitigate this situation, as an optional feature, the REI could store a blocked request from a VoIP EI, wait until the Voice Net is available, and then initiate the request to the Voice Net. If this occurs, the REI would send a busy tone immediately followed by an available tone to the EI, followed by periodic busy tones. When the Voice Net becomes available to the EI, the REI will send a burst of two available tones to the EI.

   If the Voice Net is not busy, it is considered available.

The signaling sequence for an available Voice Net, assuming out-of-band signaling is as follows:

1. The participant keys in the talk tone sequence.

2. The talk tone is sent from the originating EI to its MLSC.

3. The MLSC converts the talk tone to the UCR-standard talk tone packet.

4. The MLSC sends the talk tone packet to the DLSC to which the RBF is registered.

5. The RBF's Master DLSC sends the talk tone packet to the RBF.

6. The RBF determines that no other conference participant has access to the Voice Net.

7. The RBF sends the talk tone packet to the REI.

8. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net.

9. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net and the Voice Net sends a signal back to the VNAR indicating that the Voice Net is available.

10. The REI, within the VNAR, formats a standard "Voice Net Available" packet and sends it to the RBF.

11. The RBF performs the following functions:

    a. Sets a flag indicating that the Voice Net is busy.
    b. Sends the Voice Net Available packet to its Master DLSC.
    c. Starts a PTT timer based on a configurable parameter.
    d. Enables voice traffic from the originating EI to flow to the REI.
    e. Blocks traffic from the Voice Net to the originating EI.

12. The DLSC sends the Voice Net Available packet to the requesting EI's MLSC.

13. That LSC converts the tone to a form that is supported by the requesting EI and transmits the tone to the EI. The LSC also signals the EI to return to the media transmission mode.

14. The requesting EI creates an audio signal indicating that the Voice Net is available and reverts to the media mode of operation.

15. The conference participant can now speak. Bearer traffic will be sent from the EI to the RBF. The RBF will send the bearer packets to the conference participants and the REI. The REI will translate the bearer traffic so that the VNAR can transmit the bearer traffic to the Voice Net.

16. **[Required: VNAR, RBF, REI – Conditional: LSC or DLSC]** The PTT session shall terminate upon any of the following activities:

    a. The VoIP end user keys in the end tone. This tone is sent to the MLSC and from there to the LSC serving the RBF and from there to the RBF.

    b. A voice activity detection (VAD) device in the RBF determines that there has been no voice activity for a configurable time.

    c. The PTT timer expires.

17. **[Required: RBF]** Upon termination of the PTT session, the system shall execute the following actions:

    a. Transmit a "PTT Terminated" tone packet to the VoIP EI (via the appropriate MLSCs) for a configurable amount of time.

    b. Reset the PTT timer.

    c. Reset the Voice Net busy flag.

    d. Re-enable the transmission of bearer traffic from the Voice Net to the VoIP EI.

    e. Send a PTT Terminated packet to the REI.

    f. The REI will convert the information in the PTT Terminated packet to the form required by the VNAR to terminate the PTT session in the Voice Net.

    g. The Voice Net will become available to other parties who wish to speak.

18. **[Required: RBF]** At some configurable time before a PTT time-out, the system shall issue a "Warning" packet to inform the speaker the session is about to terminate. The Warning packet will be transmitted via the signaling path, from the system's DLSC to the conference participant's LSC to the EI.

19. **[Required: RBF]** The system shall ignore a talk tone generated by a VoIP EI that is in a PTT session.

20. **[Required: RBF]** The system shall ignore an end tone generated by a VoIP EI that is not in a PTT session.

21. **[Objective: VoIP EI]** It is desirable if the VoIP EI could be modified to include a special control key that must be depressed to maintain the emulated PTT session. This would emulate the PTT action at a radio more accurately, and potentially reduce dead time associated with the use of a timer.

22. **[Required: RBF]** The system shall provide the following configurable mechanisms to mitigate situations where the VoIP EIs might not support tone patterns to define the beginning and end of a PTT session:

    a. The conference manager shall have the ability to place the VoIP EI in listen-only mode.

    b. The system shall transmit a "warning tone" to the VoIP EI if there is voice traffic generated from a radio on the Voice Net.

    c. The system shall invoke a configurable off-on feature, which will limit the time duration of transmissions from the VoIP EI. The system shall not forward traffic from such devices for more than the configurable amount of time. If the system terminates a PTT session based on this parameter, it shall not forward traffic from the VoIP EI until a configurable amount of time has passed since the end of the last transmission period.

23. **[Required: RBF]** Table 6.1.7-1, Control Information: VNAR to VoIP EI, defines the standard tones that shall be used to convey status information to the VoIP EIs.

Table 6.1.7-1. Control Information: VNAR to VoIP EI

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
1	Voice Net Available	TBD	The Voice Net is operational and is not busy.	Caller can start to talk. Typically sent in response to a talk tone request sent by the caller (See Table 6.1.7-2).

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
2	Voice Net Operational But Not Available	TBD	The Voice Net is operational but cannot be accessed because of a lack of resources to support call; also used to indicate that the call has been preempted.	Caller should hang up and try again later. Typically sent in response to a talk tone.
3	Voice Net Busy – “Busy Tone”	TBD	The Voice Net is operational and reachable, but is busy.	Caller should try again later. Typically sent in response to a talk tone.
4	PTT Terminated	TBD	The VNAR has terminated the PTT session based on a request from the VoIP caller or a time-out.	For information purposes; caller should stop talking.
5	Voice Net Secure	TBD	Transmissions from the VNAR to the Voice Net are sent in encrypted or scrambled mode.	For information purposes. Typically sent in response to a talk tone.
6	Voice Net Plain Text	TBD	Transmissions from the VNAR to Voice Net are sent in plain text mode.	Caller should not talk if he or she were expecting that transmissions would be secure. Typically sent in response to a talk tone.
7	Stop Transmitting	TBD	Stop talking: either the speaker is talking for too long, or there is a higher priority speaker who needs access to the Voice Net.	The VNAR will block voice from reaching Voice Net. Caller should stop talking.
8	Warning	TBD	The VNAR will terminate voice traffic from VoIP EI within a configurable period.	Caller should be aware that he or she will soon get a stop transmitting signal.
LEGEND ID Identification PTT Push-to-Talk		TBD To Be Determined	VNAR	

NOTE: The typical Voice Net will not generate all the control signals shown in Table 6.1.7-1, Control Information: VNAR to VoIP EI. It is up to the designer of the VNAR to determine which control signals must be implemented. However, if a signal shown in Table 6.1.7-1 is used, the tones transmitted shall be the ones defined in Table 6.1.7-1. It is possible that future generations of TRNs will create additional signals. If required, this UCR will be modified to accommodate the new signals.

Table 6.1.7-2, Control Information: VoIP EI to VNAR, defines the signals and standard tones that shall be used at the VoIP EI to indicate the start and termination of a PTT session.

Table 6.1.7-1, Control Information: VNAR to VoIP EI, and Table 6.1.7-2, Control Information: VoIP EI to VNAR, do not include AS-SIP signaling, which is discussed in Section 5.3.2, Assured Services Requirements.

Table 6.1.7-2. Control Information: VoIP EI to VNAR

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
1	PTT Request – "Talk Tone"	TBD	A tone that indicates the start of a PTT session.	The VNAR will format and initiate PTT request to VNAR, if the Voice Net has PTT capability.
2	PTT Terminate – "End Tone"	TBD	A tone that indicates the termination of a PTT session.	The VNAR will terminate PTT request to Voice Net.

LEGEND
ID Identification TBD To Be Determined VNAR
PTT Push-to-Talk

6.1.7.5.4 Codec Translation Functional

1. **[Required: REI]** The system shall support at least one of the following RTS VoIP codecs:

   a. G.711a and µ-law (64 kbps)
   b. G.723.1 (5.3 and 6.3 kbps)
   c. G.729d (6.4 kbps)
   d. G.729e (12.4 kbps)

2. **[Required: REI]** The system shall support the Enhanced Mixed Excitation Linear Production (MELPe) codec at 2.4 kbps and lower rates.

   NOTE: When a call is set up between an RTS VoIP EI and the RBF, the codec function will negotiate the codec using the SDP. When a call is set up between an REI and the RBF, the codec function will negotiate the codec using the SDP. The codec function will attempt to minimize the bandwidth required between the RBF and the REI.

3. **[Required: RBF]** The codec function is always associated with the RBF and will reside in the same device as the RBF.

6.1.7.5.5 End Instrument Functional Changes

1. [**Conditional: AEI, PEI]** The system shall support the PTT signaling functions described in Section 6.1.7.5.3, Push-to-Talk Functional Requirements.

2. **[Objective]** It is highly desirable that the EIs be configured to directly process the standard PTT signaling packets used to create the audio tones and convey user keystrokes during a PTT session.

3. **[Conditional]** It is permissible to have the EI's MLSC create the standard signaling packets and use a nonstandard approach to conveying that signaling information between the EI and its Master LSC.

4. **[Objective]** It is highly desirable that EI PTT signaling support be implemented by software and configuration downloads, rather than hardware changes in the EIs.

6.1.7.5.6 LSC Functional Changes

[Conditional: LSC] The system shall support the PTT signaling function additions to AS-SIP described in Section 6.1.7.5.3, Push-to-Talk Functional Requirements. If the EI does not directly create and process standard PTT signaling packets, it is highly desirable that the LSCs be able to download software changes to the EIs as described in Section 6.1.7.5.5, End Instrument Functional Changes.

6.1.7.6 Network Management

General NM requirements are specified in Section 5.3.2.17, Network Management.

[Required: RBF] The system shall identify the unique name of the TRN supported by the conference.

6.1.7.7 Tactical LAN Requirements

The assured services objectives are difficult to achieve in the Tactical-edge networks (TNs), due to dynamically changing connectivity, limited bandwidth, unstable environment, and limited equipment. The TNs are often non-ASLAN compliant.

The UCR does not permit non-ASLAN-compliant devices to support special C2 and C2 users. This architecture has to be modified to allow an EI located at a non-ASLAN Tactical location to support special C2 and C2 users, provided a Type I encryption mechanism is applied to the call signaling messages and bearer traffic.

6.1.7.7.1 Physical Media Requirements

[Required: VNAR] The system must support at least one of the following Ethernet types:

- 10 Base-T
- 100 Base-T
- 1000 Base-T

6.1.7.7.2 Dial Plan and Routing Requirements

(Reference to Section 5.3.2.16, Worldwide Numbering and Dialing Plan.)

[Required] Each Voice Net shall be assigned a routable user identity, which can be one of the following: DSN number, Tel-URI, SIP-URI, and FQDN.

6.1.7.7.3 DSCP

[Required: RBF, REI, VNAR] The system shall provide a configurable mechanism to mark DSCPs in the header of IP packets. The default marking shall be as defined in Section 5.3.3, Network Infrastructure End-to-End Performance Requirements.

6.1.7.7.4 Traffic Engineering

The REI supports only voice and control traffic, and can apportion that traffic in any manner as determined by traffic engineering. The number of subscribers that need to be supported will be determined by each Program Office (PO).

The VNAR should operate within the overall network voice E2E delay and jitter requirements IAW Section 5.3, IP-Based Capabilities and Features, and voice requirements, specifically Section 5.3.1.4.1, Voice Services. The RECOMMENDED upper limit on the average post-selection delay for various E2E scenarios is defined in Table 6.1.7-3, Upper Limit on Average Post-Selection Delay.

NOTE: The UCR and REI time delays stated in Table 6.1.7-3 relate only to the time it takes to set up a call to the RBF. Typically, this is done once per conference per VoIP EI and REI. The PTT signaling requests will occur many times during the conference. The time to implement a PTT request from a VoIP EI and return an available signal to the EI involves round-trip delay between the EI and the REI. Times will vary from subseconds if there are no satellite links involved, to as many as 2–3 seconds, if satellite links are involved. The maximum delay to release an EI-initiated access to the TRN is determined by one-way delay and is proportionately less.

Table 6.1.7-3. Upper Limit on Average Post-Selection Delay

TYPE OF DELAY	UCR	REI
Local intratheater DSN call signaling during normal network traffic load	1 second	TBD
Local intrabase DSN call signaling during normal network traffic load	1.5 seconds	TBD
Worldwide DSN call signaling during normal network traffic load	6.0 seconds	TBD
Global DSN call signaling during normal network traffic load	8.0 seconds	TBD

LEGEND
DSN Defense Switched Network TBD To Be Determined UCR Unified Capabilities Requirements
REI

6.1.8 Deployed (Tactical) Master LSC and Subtended LSC Requirements and Dynamic ASAC (DASAC) Requirements in Support of Bandwidth Constrained Links

Since these requirements are applicable to the Fixed (Strategic Enterprise), as well as to the Deployed (Tactical) environment, these requirements are defined in Section 5.3.2.30, Master LSC and Subtended LSC Requirements and Dynamic ASAC (DASAC) Requirements in Support of Bandwidth Constrained Links. Many of these requirements, which are mandatory for the Deployed environment, are conditional for the Fixed environment.

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