COMMERCIALIZATION OFFICE - PILOT OPERATIONAL REQUIREMENTS DOCUMENT

Interoperable Communications Switch Operational Requirements Document (ORD)

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1 General Description of Operational Capability

As a goal, first responders would like to be able to speak to anyone at any time in any place. With the ubiquitous cell phone, that vision seems to be nearly a reality. There is a natural desire to extend this near reality to the far more complex environments of mobile platforms , remote locations (middle of the ocean, out in the desert, atop mountains), and scenes of destruction (earthquakes, explosions, fires).

While the inability to complete a cell phone call successfully may be an annoyance in a personal situation, the inability to communicate can have deadly consequences in a public safety situation. It is therefore critical that those responsible for communications in these organizations plan ahead for contingencies, set realistic expectations, acquire necessary equipment, and conduct training on a periodic basis.

Regarding expectations, it is realistic to pre-engineer multiple solutions to specific interoperability challenges that can be relied upon in an emergency. It is not realistic to think that on-the-fly personnel can expect to successfully interoperate between communications media that have not been previously analyzed and engineered for interoperability. There are numerous challenges to successful interoperability. The right combination of equipment, knowledge, and training will lead to mission critical interoperability when it's needed most. Wishful thinking and ignoring the complexities will, in contrast, provide a false sense of security and lead to failure.

The problems and issues associated with different radio systems not being able to communicate with each other have been known to first responders for many years. The communications problems surrounding the terrorist attacks on September 11, 2001 significantly raised the visibility of this issue and have led to numerous and varied attempts to improve communications interoperability amongst first responders.

1.1 Capability Gap

One primary method of resolving communications interoperability is having all involved parties using the same, or at least interoperable, radios, whether they are cellular, portable, fixed or mounted. Since many first responders have already invested significantly in their current radio systems, acquiring new radios is often not a practical solution. This leads to the second means of resolving interoperability issues, the use of a gateway or switching type of device or system that can quickly and easily connect two or more otherwise non-interoperable radio systems. This system would allow multiple first responders to talk to each other either directly or via radio nets, all while using their existing radios, cell phones, or telephones.

1.2 Overall Mission Area Description

The mission area covered by this ORD is all public safety related events where first responders must communicate with other first responders using communications media such as radios and telephones that are not normally interoperable. This includes different agencies and types of first responders (police, fire, EMTs, etc.) and first responders from different jurisdictions and/or locations (city, county, state, federal, etc.)

1.3 Description of the Proposed System

Responders in the field need access to a switching system with the capability to integrate voice communications of all types in a special evolution or command and control type environment such as an emergency operations center (EOC). The proposed interoperability switching system will provide the user the ability to provide advanced private branch exchange like capabilities between handsets connected through PSTN, IP, local radio systems (e.g. Land Mobile Radio (UHF or VHF)), commercial wireless (cellular/PCS and satellite) and other standard interface systems. The switching system shall include a full range of switching functions for telephony, radio circuits, simultaneous plain and P-25 encrypted circuits, progressive radio and telephone conferencing and netting, extensive administrative support for configuration planning and event and call logging, and a wide variety of system interfaces. It will support a wide range of commercial voice terminals (analog and digital), radios, wireless systems (such as IP-DECT), integrated voice communication terminals, assignable loudspeakers, and virtually any other analog or digital voice source. The switching system will be able to provide interoperability on a much broader scale than simply tying together radio nets. The switching infrastructure must able to bring all the types of voice communications needed by each user together in a single voice terminal. For some, a telephone is sufficient; for others, a multi-purpose integrated terminal that can handle both radio and telephone calls is appropriate.

The switching system shall be capable of connecting to all types, brands and styles of first responder land-mobile radios in a fixed or mobile communications center. The system will be the hub that connects or networks different types of radios (even radios on different frequency bands) and at the same time allows the local users at the communications center to join multiple radio networks and communicate over telephones, intercom and landlines, all at the same time. Figure 1.3-1 shows the critical functions and interfaces of such a system that will allow first responders, and anyone else associated with a particular emergency operations or communications center, to communicate with one another while using different systems.

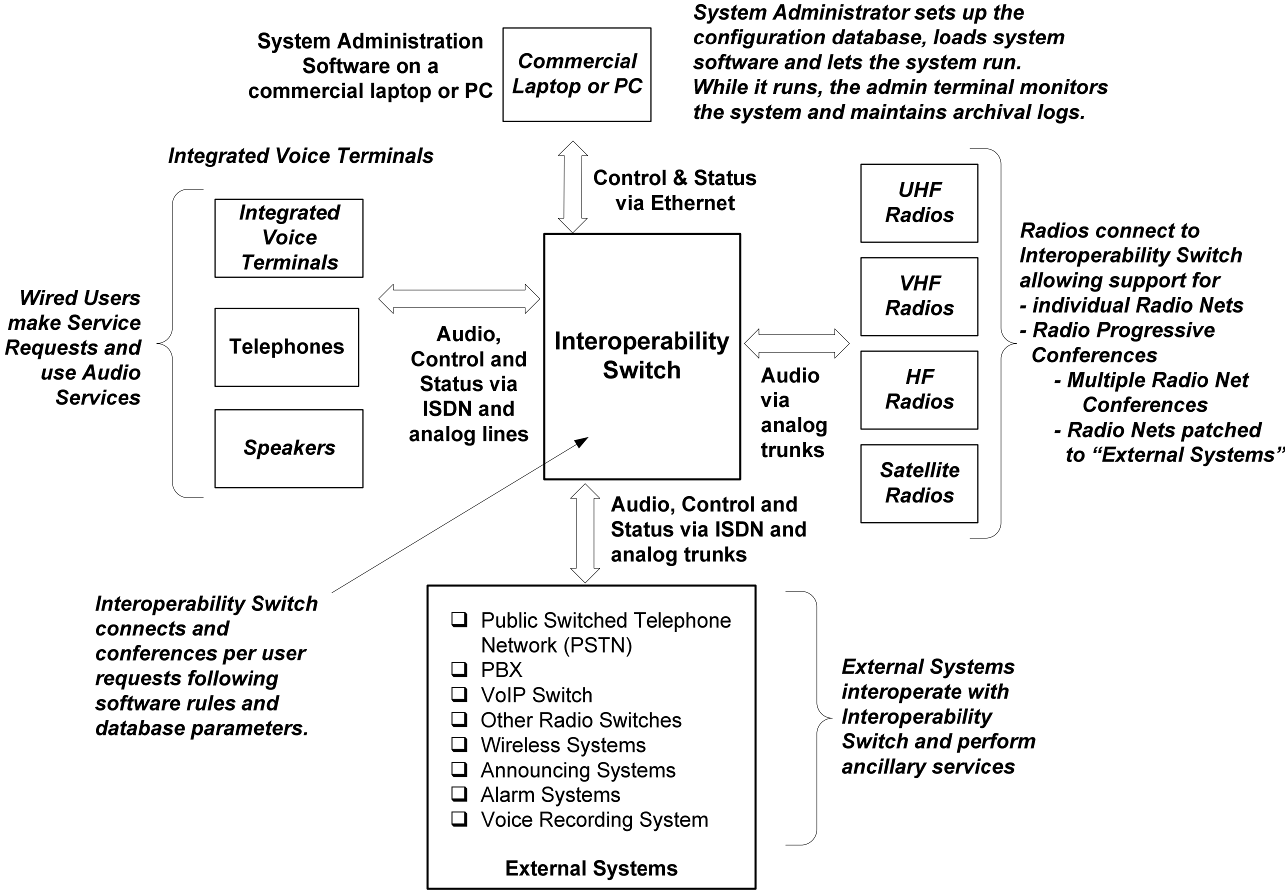

Figure 1.3-1 The critical functions and interfaces of an interoperability switching system to provide first responders interoperability with each other and the rest of the world

1.4 Supporting Analysis

This ORD is supported by analysis done by DHS S&T.

1.5 Mission the Proposed System Will Accomplish

The proposed system will be able to connect and network different and various types of radios, wireless systems, integrated voice terminals, telephones and other communications media such as PBXs, VoIP Switches and the Public Switched Telephone Network (PSTN). Integrated voice terminals are defined as devices that can handle several functions, such as radio calls, telephony, and intercom simultaneously. The proposed system will provide a means for users (first responders and those that need to talk to them) with different communications devices and media to seamlessly communicate and interoperate with one another.

1.6 Operational and Support Concept.

1.6.1 Concept of Operations

The communications interoperability switch will enable first responders to communicate with each other and with communications center personnel using different types of radios, cell phones, telephones and other communications means. This system will integrate voice communications so that police, fire, EMT personnel of all types and from all jurisdictions will be able to easily talk to each other using whatever means of communications they have. Figure 1.6-1 shows the concept of first responders using various devices all connected to the Interoperability Switch by either wire or wireless, being able to communicate with one another. This communications can be either conferenced, networked (netted) or point to point. The proposed system will typically be located in a fixed communications or command center such as an emergency operations center (EOC) but will also be sized to be able to be located in a mobile station if needed.

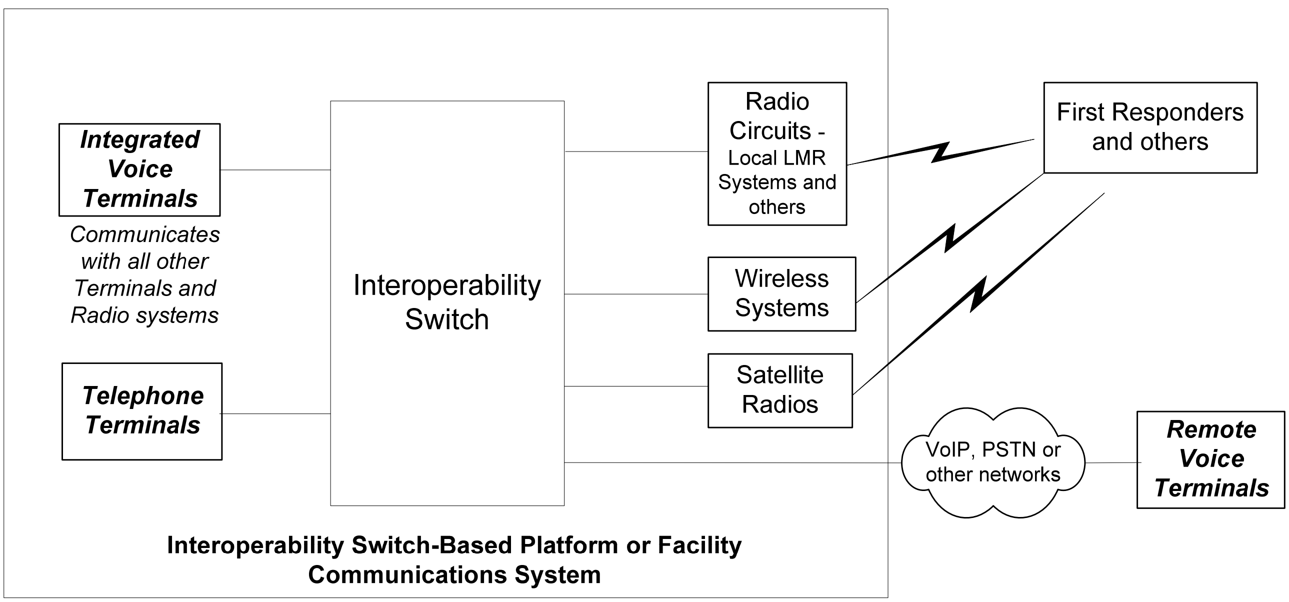

Figure 1.6-1 An Interoperability Switch-Based Facility Communications System Provides Networked Communications Between any Number of Agencies and Personnel

The proposed system will provide the following operational capabilities:

1. Enable all agencies and entities to keep their existing radios and other voice terminals, yet integrate them together in a system-of-systems (SoS).
2. Provide the ability for communications operators to quickly and seamlessly connect or disconnect any number of first responders with a few button pushes (no laptop needed).
3. Deliver calls without blocking.
4. Enable managers, and other authorized users, to monitor as many communication channels or circuits as they require (or can personally handle) to achieve maximum situational awareness.

5. Interface with security and encryption, if required, to provide transmission security, and easily control who can hear which conversations.

1.6.2 Support Concept

The proposed system shall be maintainable either by the equipment provider or by personnel trained to maintain the system.

The design of the proposed system shall support easy installation by the equipment provider or other trained personnel. Some knowledge of the (fixed or mobile) emergency operation center's interfaces (such as radios, telephones and power) will be required in order to plan and do the installation.

Maintenance requirements for the system shall be minimal. Each unit shall include basic self-test mechanisms to indicate proper operation. System design shall allow for easy replacement of a defective line replaceable unit (LRU) by a new unit with no need for user level repair maintenance. Defective LRUs will be returned to the manufacturer for disposition.

Spare parts will be made available by the equipment provider if not available as a commercial-off-the-shelf (COTS) item.

Training shall be provided by the equipment provider to either a system trainer (via a train-the-trainers session) or to the users and operators at the installed site at a time convenient to the users and operators.

2 Threat

The proposed system counters any threat potentially caused or exacerbated by first responders not being able to communicate with one another. In critical situations, the inability of responders to be able to communicate with one another or with command and control authorities could cause loss of life. The interoperability provided by this system will eliminate communications breakdown or failure as a source of issues when dealing with the threat or situation.

3 Existing System Shortfalls

Existing systems that provide interoperability have the following weaknesses:

- The number of devices and nets supported is inadequate to serve as a radio switch for all but the very smallest of applications.
- No support is provided for integrated voice terminals. Integrated voice terminals greatly improve the mission effectiveness of users through:
  - Allowing multiple circuits (radio or telephony) to be monitored simultaneously while supporting one channel in active mode.
  - Provision of dynamic key text and color to make communications intuitive and responsive to the specific needs of the user.
  - Supporting advanced interaction with remotely controllable radio terminals. Such interaction requires an intelligent switch.

- Allowing a member of a conference (using an integrated voice terminal - IVT) to monitor the terminal traffic and dynamically manage the conference by adding members or dropping others out of the conference as circumstances warrant.

- No support is provided for secure voice circuits. Even if secure conferences are not attempted (with multiple radios and encryption devices), these applications do require the switching system to support secure radio circuits at the same time that plain radio circuits are operating. This imposes requirements on the switching equipment that the current systems do not support.

- While the Human-Machine Interface (HMI) for current systems may be adequate for the duration of a specific interoperability net, it is not acceptable for a general radio switch.

  - The HMI for a current system is accomplished using a laptop. Thus, the management of any conference requires someone to use a central gateway laptop for conference setup and management. This requires personnel resources that will not be necessary when each conference can be managed at the voice terminal of the leader of the conference.

  - The ideal managers of specific conferences are likely to be different individuals depending upon the mission served by the conference. Thus, a central laptop is much less effective for HMI than allowing integrated voice terminals to serve the conference managers as needed operationally.

4 Capabilities Required

4.1 Operational Performance Parameters

The proposed system shall provide required voice and control signal connections to support terminal-to-terminal calls, terminal to net calls, external system calls to terminals and nets, and combinations of these.

The proposed system shall provide a “non-blocking” architecture such that calls cannot be blocked because of switch limitations.

The proposed system shall support ISDN and POTS lines and trunks, and provide for non-blocking traffic flow among all switch port connections, for up to 2000 subscribers (configuration dependent).

The proposed system shall be able to provide connections for three classes of terminal devices:

- Direct BRI S/T line connections for user terminals such as integrated voice terminals and ISDN phones;
- Direct POTS connections for POTS and Analog phones and connections;
- Network Termination (NT) adapters for converting between the BRI S/T lines and special analog interface connections such as radios and PA systems.

The proposed system shall provide a primary rate interface (PRI) trunk connection for interfacing to private branch exchange (PBX) systems and radio communication systems (RCS).

The proposed system shall be capable of providing redundancies to ensure protection against single-point failures.

The proposed system shall support full-duplex connections, conferencing, self-test operations, and both plain (unencrypted) and secure modes of operation between designated terminals and systems.

The proposed system shall be created such that users from outside the EOC's area of responsibility are able to communicate with local first responders.

Digital Terminals

The proposed system shall support digital/ISDN terminal direct dial service to other dial terminals and direct dial access (when properly class marked) to nets and external systems that interface with the Interoperability Switch.

Specific communications interoperability switch features available for use by digital terminals will be limited only by the physical configuration of the terminal and the accesses or class marks available to it.

Digital/ISDN terminals shall provide an interface to an associated Interoperability Switch in accordance with industry standard digital BRI S/T characteristics and requirements, and to standard accessory connections associated with the terminal (e.g., handsets, headsets, speaker extensions).

Integrated voice terminals (IVT) are multi-functional ISDN terminals that will have push-to-talk capability and can therefore make radio calls in addition to making standard telephone and intercom calls.

The communications interoperability switch shall provide the power for the ISDN terminals.

Analog Terminals

The communications interoperability switch shall support analog POTS (plain old telephone service) services that operate with a standard loop start signaling interface, for connections to POTS terminals, associated FAX machines and external connections that appear to the POTS interface as a terminal.

Analog POTS terminals shall provide an interface to an associated Interoperability Switch in accordance with the requirements of industry standard EIA/TIA-464B, and to standard accessory connections for the terminal (e.g., handsets).

The communications interoperability switch shall provide the power for the analog terminals.

System Features

The proposed switching system shall provide the connection paths for the voice and control signals transmitted and received by dial terminals and net terminals. The types of call connections that shall be provided are as follows:

- Dial terminal to dial terminal calls: The calling party activates the dial terminal, receives a dial tone or indication and presses or selects (keys) the appropriate buttons on the terminal for the desired service
- Dial terminal to Net terminal calls: The calling party initiates the call and keys the terminal for the desired service. If the dialed number or single button access represents a net, the calling party will be connected to the net.
- Designated terminal to External System interface connections (such as PBXs, VoIP Switches or the PSTN)

The proposed system shall provide and support the services and features shown in Table 4.1-1. The paragraphs that follow the table define the service requirements in additional detail.

Table 4.1-1 Matrix for Required Types of Terminal Calls Operations and Services

	CALL OPERATION OR SERVICE	TERMINAL APPLICABILITY Integrated Voice Terminal	ISDN COTS Terminal	POTS or Analog Terminal
1	Call Hold	X	X	X
2	Call Transfer	X	X	X
3	Abbreviated Addressing	X	X	X
4	Progressive Conference	X	X	X
5	Preset Conferencing	X	X	X
6	Meet-me Net	X	X	X
7	Privacy/Auto Override	X	X	X
8	Call Forwarding	X	X	X
9	Call Waiting	X	X	X
10	Assigned Access	X	X	X
11	Access Restriction	X	X	X
12	Alternates	X	X	X
13	Plain or Secure Calls	Either	Plain	Plain
14	Call Monitor (Simultaneous)	X	-	-
15	Push To Talk	X	X	X
16	Intercom Announcing	X	-	-
17	Intercom Hotline	X	-	-
18	Emergency Reporting	X	X	X
19	Speed Calling Lists	X	X	X
20	Call Groups	X	X	X
21	Discriminating Ringing	X	X	X
22	Caller ID	-	X	
23	Activity Detection	X	-	-
24	Analog connection	X	X	X
25	PA Announcing System connection	X	X	X
26	Alarm System Connection	X	-	-
27	Radio Net access	X	X	X
28	Radio Progressive Conferencing	X	X*	X*
29	Assignable Speaker	X	-	-
30	Voice Recorder – Record	X	-	-
31	Voice Recorder – Playback	X	-	-

X Required
- Not Required
* Does not need to initiate a Radio Progressive Conference, but can be added by an Integrated Voice Terminal

System Call Processing Requirements

The following subparagraphs of this section provide a brief description of Call Processing types and services for the communications interoperability switch (shown in Table 4.1-1).

Call Types

Call Hold

Call Hold places an engaged call on hold to allow a subscriber to consult a third party. A Call Hold capability shall be available to all communications interoperability switch subscribers who are involved in a two party call.

Call Transfer
Call transfer provides a capability to transfer a received call to another terminal, and also permit three-party calls.

A call transfer capability shall be available to all Interoperability Switch subscribers who are involved in a two party call.

Call transfer shall refer to both a "blind transfer" (transferring party hangs up before the transfer is answered) and an "active transfer" (transferring party waits for the transfer to be answered before completing the transfer). Active transfer is also known as a transfer with introduction.

Call transfers to PSTN Lines, nets, conferences, and multi-party calls shall not be allowed. Additionally, call transfers from nets and conferences shall not be allowed.

Subscribers currently connected to nets or in conferences shall not have the capability of call transfer.

Abbreviated Addressing (Speed Dialing)
Abbreviated Addressing / Speed Dialing permits designated dial terminals the capability to use abbreviated addresses for dialing. Entering a designated abbreviated addressing code into a terminal keyboard (typically two digits preceded by an "asterisk") shall initiate a call from the dial terminal.

Speed dial numbers shall be programmable at both the Local level (Speed Dialing numbers that are applied to a unique terminal) and at the Global level (Speed Dialing numbers that are applied to all terminals). Each Local Level Speed Calling List is unique to a specific terminal while the Global Level is available to all configured terminals.

The system administration terminal software (SAT) shall allow for the configuration of up to 10 Local Level Speed dial numbers per terminal, and the SAT shall allow for the configuration of up to 80 (T) global level speed dial numbers.

Each integrated voice terminal shall provide the ability to program up to 20 (T)/25 (0) pre-programmable dial keys or buttons, local to the integrated voice terminal, that are to be used for speed dial. Additionally most ISDN telephone terminals provide the ability to program speed dial keys available on the terminals.

Privacy/Automatic Override
It shall be possible to assign a privacy override capability so that the “busy” condition of a called dial terminal, and Call Waiting if applicable, can be overridden by someone with the proper authority. This feature will allow selected users to exercise preemption capabilities to cut into or override terminals being used for calls with lower precedence levels. Two methods shall be available for initiating Privacy Override in designated terminals:

a. After receiving dial tone, the subscriber depresses the # key and then keys in the called terminal directory number; or
b. After keying the called terminal directory number and receiving busy, the subscriber depresses the # key within three seconds after receiving busy tone.

A one-second override tone shall be placed on the existing connection, such that all members of the connection hear the tone, before connecting the override call.

An overridden terminal with the call waiting capability that is active on one call appearance shall have the previously active call placed on hold.

Call Forwarding
Dial terminals designated or class marked for call forwarding shall be able to have all incoming calls routed to another dial terminal, through subscriber implementation.

Three types of call forwarding shall be available:
a) Unconditional, where calls will be automatically rerouted;
b) Call forwarding busy, which reroutes an incoming call only if the called terminal is busy;
c) Call forwarding no reply, which reroutes an incoming call if there is no answer within a specified amount of time.

To implement call forwarding, the subscriber shall dial a configurable special service code appropriate to the type of call forwarding, followed by the four-digit number of the terminal to which the calls are to be forwarded.

Upon the completion of a terminal call forwarding to a valid terminal, the subscriber shall be notified with a confirmation tone.

To cancel call forwarding on a terminal, the subscriber shall dial the configurable special service code assigned for cancelling call forwarding.

Call Waiting
A call waiting capability shall be available for designated terminals that provide a visual and/or audible indication at a terminal engaged in an established call, to alert it that an incoming call is awaiting connection. A single user action at the designated terminal shall place the engaged call on hold and connect to the waiting call.

Assigned Access
It shall be possible for selected dial terminals to have an assigned access (by class of service) to any combination of the following: individual nets, public address systems, radio trunks, and PSTN connections.

Terminals assigned such access shall be able to obtain the desired connection by keying the appropriate number from the address numbering plan, and terminals that attempt to complete a call to a destination to which access has not been assigned will receive an unavailable tone.

Access/Class Mark Restrictions
It shall be possible to assign access restriction (class mark) categories to all Interoperability Switch line connections, circuits and terminals for the purpose of controlling intercommunications to or between them. Class marks (CM) provide a means for software to control user accesses and privileges (such as call waiting, call forward, and override).

An assigned or default class mark shall apply for each terminal, circuit or call feature so that if the CM appears within the class of service (COS) restricted category for a calling party (CLG) terminal, the CLG terminal will be prevented from connecting to the called terminal, circuit or call feature. COS and CM assignments for individual terminals will be provided from the SAT (via the communications interoperability switch).

Alternates
It shall be possible to designate three alternate terminals to be tested in the event that the primary terminal is busy, unavailable or idle, for a minimum of 16 (T) / 32 (O) dial terminals.

If the primary terminal is busy or unavailable when called, the alternate terminals shall be checked in order and the first idle alternate rung.

If an idle alternate is rung and not answered before the ring period timeout, the next alternate terminal shall be rung.

If the last alternate is idle and not answered a calling ISDN terminal will be placed on-hook while a POTS terminal shall receive unavailable tone.

If the dialed terminal and all alternates are busy, the calling party shall receive busy tone.

If the dialed terminal and all alternates are busy and the caller chooses to override within 3 seconds of receiving busy tone, the dialed terminal shall be overridden.

Call Groups
The Interoperability Switch shall support a telephone call groups' capability, for:
a) Rotary hunting (where an incoming call is automatically rerouted to another terminal in a Call group if the first terminal is busy, unavailable, or is not answered during the ring time out period.
b) Call pickup within a call group (where any terminal in a call group can pick up a ringing call to a group member, by dialing a designated call pickup number), for at least 16 (T) / 32 (O) groups with a minimum of 16 (T) / 20 (O) subscriber members per group.

Plain or Secure Calls
Controls for integrated voice terminals only shall be provided to permit calls in both plain and secure modes of operation.

When a circuit transitions to secure mode all plain-only ports connected to the secure circuit shall have their audio reception blocked until the circuit transitions back to plain mode.

Transmission of plain-only ports shall still occur to the secure circuit. The communications interoperability switch will be responsible for security by configuring, connecting, tracking, and disconnecting circuits. When an incompatible security connection is attempted, the integrated voice terminal shall display a security mismatch with a security mode indication on the display.

An integrated voice terminal shall not have the capability to change the security mode of a call while its PTT is depressed or while the PTT of a terminal connected to the circuit is depressed.

When a radio net is switched to secure mode, all plain-only terminals in the net shall:
- Be disconnected from the net.

- Receive a security mismatch (Unavailable) tone.

If a plain only terminal attempts to override a terminal with at least one call appearance in a secure radio net, the following shall occur:

- The override is unsuccessful and there is no disturbance to the net.
- The plain only terminal gets Unavailable tone.

Call Monitor

A call monitor capability shall be supported with integrated voice terminals that permit an integrated voice terminal with an existing call connection to accept or originate a new call connection without disconnecting the existing call. The first key or button pressed in accepting or originating a call will move an existing call into the monitor mode, where it is held and monitored while the user participates in the new call.

The first key pressed in accepting or originating a call shall move the existing call into the monitor mode on the ISDN Bearer 2 channel.

The integrated voice terminal shall be able to monitor calls on the Bearer 2 channel while the user participates in an active call on the Bearer 1 channel.

Discriminating Ringing

The communications interoperability switch shall support a discriminating ringing capability, by providing a user selected discriminating ringing for calls originating within the system, originating outside the system (PSTN), or from interface connections (e.g. wireless system).

Caller ID

The communications interoperability switch shall provide a calling line identification capability (Caller ID) on all ISDN terminals equipped with a user display (reference ANSI T1.625 as a guide).

Activity Detection

The communications interoperability switch shall provide an activity detect call feature which provides an integrated voice terminal user a visual indication of voice activity on a monitor channel.

The operator shall be provided the ability to toggle this feature on and off from the integrated voice terminal.

When enabled, only the integrated voice terminal keys or buttons that are occupied with calls in monitor mode (illuminated amber) shall blink when audio is being received on the channel associated with the key. This makes it possible for the user to be active in one call while knowing exactly where the monitor audio in the speaker is originating.

When this feature is disabled, monitor calls shall remain solid amber even when audio is being received.

Conferences and Nets

Progressive Conference

For a subscriber terminal that is properly class marked, it shall be possible to set up a full-duplex Progressive Conference capability, whereby terminals are called to join a conference.

A minimum of 15 (T) / 20 (O) progressive conferences in progress or in setup at one time shall be allowed, for 12 (T) / 14 (O) conferees per conference. Setup of a conference will be initiated by a conference originator, and add-on permitted by any conference member with the proper permissions (the members Class of Service is not restricted from performing a progressive conference)

Preset Conference (and Command Net Call)
A preset conference is a call between a set number of previously designated terminals. At least 15 (T) / 20 (O) preset conferences of 12 (T) / 15 terminals each shall be supported.

Dialing the preset conference directory number from one of the designated terminals shall ring the other designated terminals.

Each designated terminal (of a predefined conference member) shall be added to the preset conference if it goes off-hook before the end of the ring period, which shall be programmable up to a maximum of 45 seconds.

Command net call is similar to a preset conference except that it does not allow automatic privacy override.

At least 15 (T) / 20 Command Nets of 12 (T) / 15 terminals each shall be supported.

Meet-Me (Voice) Net
A meet-me net capability shall be provided, whereby participating terminals are not pre-assigned to the net but will enter it with a single action depression (on a integrated voice terminal) or defined programmable directory number with no additional user action.

Dialing a defined meet-me number shall immediately connect a terminal to the meet-me net.

Every terminal that dials the meet-me net directory number shall be connected into the net with the ability to disconnect and reconnect without disturbing other net participants.

Each net shall support a capacity of at least 12 (T) / 15 (O) participants. The minimum simultaneous net capacity shall be at least 15 (T) / 20 (O) nets.

Emergency Nets/Calls
An emergency reporting net capability of up to 3 (T) / 4 (O) nets shall be provided to receive emergency calls from any dial terminal, with one terminal assigned to each emergency net for handling incoming emergency calls on that net, and identified as the responsible dial terminal (RDT).

When a called RDT goes off-hook, it will be connected to its emergency net, and any subsequent calls to the emergency number or associated net number will be connected to the emergency net and be able to converse with other net members.

Emergency reporting shall be possible for each of five 'readiness' conditions, and under each condition of readiness a particular RDT may be designated as responsible for handling emergency calls on one or more emergency reporting nets.

An emergency reporting net shall be identified by up to two emergency telephone numbers (i.e., 2211 and 911) in addition to a net number.

The following call/connection procedures shall be implemented:
Any terminal calling the emergency number and RDT is not-busy, shall receive a ring-back tone until the RDT operator goes off-hook (or integrated voice terminal equivalent), at which time both parties shall be connected to the corresponding emergency net.

Subsequent callers calling the emergency number or the emergency net number shall be connected to the corresponding emergency net and be able to converse with other net members.

If the call to the RDT cannot be completed due to equipment problems or settings, the operator of the calling terminal shall receive an unavailable tone.

If an emergency call is made to the RDT while it is busy on a call to other than its assigned emergency net, all parties on the existing call shall hear a one second emergency tone added to their conversation in progress, and then will be placed on hold while the RDT is automatically connected to the Emergency Net.

The RDT shall be overridden by an emergency net call even if the RDT is currently on a non-overridable call on its non-emergency number. The RDT operator may then retrieve any of the parties on hold.

The RDT shall continue to be connected to the corresponding emergency reporting net even if the calling terminal should go on-hook.

The RDT’s connection to the net shall be broken only when the RDT goes on hook or deactivates.

At least 3 (T) / 4 (O) Emergency Nets of 12 (T) / 15 (O) terminal participants each shall be supported.

Address Numbering Plan
An address numbering plan capability will be provided that permits each terminal, net, interface channel or service code to be identified by a discrete four-digit number. The address numbers are used in switch service operations for identification purposes and by the subscriber for service requests.

A numbering plan will typically be divided into two parts: a fixed or reserved set of numbers, and a directory set of numbers.

PTT and Intercom
Push-to-Talk (PTT)
A push-to-talk (PTT) capability shall be supported for integrated voice terminal connections and radio mode calls.

A voice operated transmission (VOX) PTT shall be implemented for POTS and BRI/ST Interface Boards.

Intercom Announce

Intercom capability shall be supported for integrated voice terminals, as a dedicated non-blocking service feature that establishes a talk-back connection between designated terminal users. In Intercom Announce the calling integrated voice terminal alerts the called subscriber with an audible tone. An integrated voice terminal permits a called party to hear the calling party even if the called integrated voice terminal is busy, and a single action at the called integrated voice terminal establishes a connection in the reverse direction to permit the called party to talk to the calling party.

The initiator of the IC call shall have an immediate half-duplex connection to the monitor channel of the other integrated voice terminals in the IC group. The other integrated voice terminals will hear the originator without any action on their part.

An integrated voice terminal key in the IC ringing state shall beep and continue flashing until answered or the caller disconnects.

IC ringing shall not time out. If not answered, the call shall remain in the ringing state until the calling party disconnects.

Pressing the IC key or button on a called integrated voice terminal shall establish full-duplex audio between the terminal, the initiating terminal and any other integrated voice terminals that have answered.

If other members disconnect, leaving one remaining member, the call shall remain active.

An integrated voice terminal shall have the ability to leave the IC call and re-enter the call by depressing the IC key.

An integrated voice terminal operator who presses the IC key to return to an active IC call shall be immediately connected.

At least 15 (T) / 20 (O) total Intercom circuits of 12 (T) / 15 (O) participants each shall be supported.

Auto Answer

Applicable to ISDN terminals with auto answer capability, an incoming ring signal shall automatically activate the terminal if its Mode switch is set to “auto-answer,” allowing the terminal to ring once and the calling party to start speaking.

The integrated voice terminal shall include a locally enabled auto-answer feature, whereby the terminal automatically answers incoming telephone and Intercom Announce calls without any user action required.

External Connection Calls

A capability shall be provided to permit dial terminals that are appropriately class marked to dial a connection to an interfacing external system, as described in the paragraphs that follow (such as to a public address (PA) system, radio net, or access to a PSTN trunk using a dialed access code).

Public Address (PA) and Alarm System Connection
A capability shall be provided for connecting to a PA or alarm system from designated voice terminals, by keying (dialing, with PTT) a designated PA or alarm system termination number.

At least 8 (T) / 12 (O) total PA or alarm system nets of 12 (T) / 15 (O) participants at least each shall be supported.

Radio Net Connection
The communications interoperability switch shall provide a radio, analog NT interface capability (application dependent) that permits a secure mode connection via the communications interoperability switch, from an integrated voice terminal to a site-provided voice radio device. This NT circuit shall present an interface that consists of BRI S/T-to-analog converter circuits and discrete control lines, for an appropriate radio channel connection that has a standardized interface.

At least 15 (T) / 20 (O) total Radio Nets of 12 (T) / 15 (O) participants at least each shall be supported.

PSTN Connection
A capability shall be provided for accessing PSTN trunks from dial terminals and integrated voice terminals that are appropriately class marked, by dialing an access code. The PSTN side shall provide the required dial tone.

Traffic Handling Capabilities
Traffic handling capabilities for the communications interoperability switch will have minimum (threshold) baseline characteristics as specified in the paragraphs that follow:

a) Traffic load and distribution - During the busiest hour the communications interoperability switch shall be capable of handling: a) 0.004 terminal-to-terminal calls originated per dial terminal per second (equates to one new call per terminal every 4 minutes), with an average holding time of 30 seconds; and b) 0.002 line-to-net calls originated per dial terminal per second (approximately one new call every 8 minutes), with an average holding time of 2 minutes. It is assumed that the percentage of these calls completed within the originating node is equal to 100% divided by the number of nodes, and that the traffic load imbalance between multiple nodes does not exceed 1.5 to 1.
b) Call Busy Factor Adjustment - A call busy factor of 25 percent is assumed, to reflect the number of dial terminals unable to make or receive calls because the line is occupied with a previously established call.
c) Call Initiation Delay - The busy hour call initiation delay measured from call initiation to receipt of dial tone shall be less than 3.0 seconds.
d) Call Completion Delay - The busy hour call completion delay measured from the last digit dialed to ring forward shall be less than 0.5 second for calls at one node, or less than 2.5 seconds for calls between nodes.
e) Blocking - An Interoperability Switch shall provide a traffic handling capability of less than one call in one thousand lost or blocked (equates to a call not going through) as a result of: an error in the controller, or a false trunk, switch or station signal.

f) Misrouting - For security requirements, the probability of call misrouting (call sent to another terminal) due to an Interoperability Switch error shall be less than one in 10^6.

Radio Progressive Conference

Scope

The radio progressive conference (RPC) feature provides a means to establish a true two-way conference call between multiple radios, integrated voice terminals, and other terminals.

Operational Concept

This feature enables an integrated voice terminal user to join two or more radio nets together to form one large net. As an example, a VHF link from one land-based agency to a helicopter could be joined to a UHF link from the same agency back to other agencies in the area. The extended network would be half-duplex, but participants on the VHF and UHF links can all hear transmissions and transmit on either link. This represents a concatenation of two nets.

In addition, the feature can be used to bring another terminal into a radio net. For example, the originator may be participating in a law enforcement UHF net and decide that someone on another IVT needs to join the conversation. That operator can call the other IVT and then conference that IVT into the radio progressive conference.

The term progressive in the title implies that additional members (Radio Nets or terminals) may be progressively added (or dropped) one at a time. These conferences can also be referred to as ad hoc conferences.

RPC Requirements

The proposed switching system shall provide radio progressive conferencing with 15 (T) / 20 (O) Preset Conferences. Each preset conference shall support at least 12 (T) / 15 (O) terminals.

The SAT shall have the capability to configure the radio progressive conference feature for any integrated voice terminal.

If a radio net or a terminal is already involved in a radio progressive conference, attempting to conference that radio net or terminal shall result in an unavailable tone at the attempting integrated voice terminal.

Assignable Speaker/Voice Recorder (AS/VR)

The assignable speaker/voice recorder (AS/VR) feature of the proposed system shall enable a user to assign speakers or a voice recorder to an interoperability switch radio net, public address net or voice net for monitoring and recording purposes.

The proposed system shall be able to interface with a public address announcing system using industry standard interfaces.

The proposed system shall be able to interface with an alarm system using industry standard interfaces.

The proposed system shall be able to interface to a voice recording device using industry standard interfaces, for the purposes of recording any of the circuits or calls that are routed through the switch.

The voice recorder's record port shall be able to be connected to a net (via the communications interoperability switch) such that all voice transmission on the net is recorded.

The voice recorder's playback port shall be able to be connected to a net (via the communications interoperability switch) such that multiple integrated voice terminals and dial terminals can listen to the playback audio.

The connection of the speaker and/or the voice recorder to a net (via the Interoperability Switch) shall be configurable from the SAT (offline or online) or from the integrated voice terminal.

4.2 Key Performance Parameters (KPPs)

4.2.1 Connectivity

The interoperability switching system shall provide at least:

Connectivity to radios 16 (T) / 32 (O)

Connectivity to integrated voice terminals - 24 (T) / 48 (O)

Connectivity to telephones 16 (T) / 32 (O)

Connectivity to wireless systems - 4 (T) / 8 (O)

Connectivity to public switched telephone networks - 1 (T) / 2 (O)

Connectivity to recording systems - 2 (T) / 3 (O)

4.3 System Performance.

4.3.1 Mission Scenarios

The interoperability switching system will typically be located at fixed area or mobile communications centers that handle emergency events such as an emergency operation center (EOC). Systems will be installed and can be up and in operation at all times in order to minimize the time needed to establish communications in the event of an emergency.

4.3.2 System Performance Parameters

The interoperability switching system shall provide at least:

*Connectivity to radios 16 (T) / 32 (O)

*Connectivity to integrated voice terminals - 24 (T) / 48 (O)

*Connectivity to telephones 16 (T) / 32 (O)

*Connectivity to wireless systems - 4 (T) / 8 (O)

Connectivity to other switches via a PRI interface - 1 (T) / 2 (O)

*Connectivity to public switched telephone networks - 1 (T) / 2 (O)

Connectivity to public address systems - 2 (T) / 4 (O)

Connectivity to other interoperability switches via a trunk - 1 (T) / 2 (O)

*Connectivity to recording systems - 2 (T) / 3 (O)

Connectivity to voice over IP (VoIP) systems - 1 (T) / 2 (O)

4.3.3 Interoperability

The communications interoperability switch will be able to interface to all radios, wireless systems, integrated voice terminals, telephones, PBXs, VoIP Switches, PA systems, recording devices and other communications media that utilize industry standard interfaces.

4.3.4 Human Interface Requirements

An integrated voice terminal (IVT) will be the primary and most functional human machine interface (HMI) connected to the communications interoperability switch for connecting and establishing radio/wireless and telephone calls, circuits, conferences and nets.

Analog and digital telephones (also known as dial terminals) will be additional HMI devices connected to the communications interoperability switch for the purpose of making and receiving calls and connecting to conferences and nets.

A system administration terminal (SAT) will act as the HMI for system configuration data entry, system configuration reports, system status reports and failure interrogation.

The SAT can be either continuously connected to the communications interoperability switch for permanent ongoing system status reporting, or be capable of being placed in offline mode during user absence or for configuration database updating (for a later database transfer to the communication interoperability switch).

When the SAT is not online, communications interoperability switch status and failure events shall be stored in the communications interoperability switch for batch transfer to the SAT when it is returned to online status.

A SAT connection shall be able to interface to a local or networked printer, if part of the configuration, for hardcopy printouts of system status.

The SAT can be any PC which is operable from 115 VAC, is available with back-up battery option, provides printer and Ethernet interface connections, is capable of running the communications interoperability switch SAT software under Microsoft Windows®.

The SAT shall provide for communications interoperability switch setup and management and for initiating switch built-in-test (BIT) operations.

The SAT shall provide a status screen displaying the latest status of the communications interoperability switch.

The SAT status screen shall contain the communications interoperability switch call and fault logs.

The SAT shall provide the user a capability to manage the system tests and view the status of the tests.

Accepted industry standards shall be applied as guidance for human engineering design criteria in the design of the proposed system, to achieve safe, reliable and effective performance by operator, supervisor and maintenance personnel, and to minimize personnel skill requirements and training time.

4.3.5 Logistics and Readiness

The proposed system is required to be operational for several days of continuous operation without interruption. No user level maintenance or spare part replacement is required. Spare PWAs should be available in case replacement is required.

Mean time between failures (MTBF) shall be 1,500 hours (T) 1,800 hours (O)

System availability (A_i) requirement shall be 0.999995 (T), 0.999997 (O) based on the following formula:

$$A_i = \frac{MTBF}{MTBF + MTTR}$$

4.3.6 Other System Characteristics

Design drivers are the interfaces and the ability of the proposed communications interoperability switch to interface to all types of radios, wireless systems, telephones, and other communications media.

Cost drivers are the interface cards for the many and varied systems to be connected to the proposed system.

Risk drivers are the ability of the communications interoperability switch to interface with many and varied different systems using readily available off the shelf interface boards without the need of designing or building new boards

5 System Support

5.1 Maintenance

The proposed system shall be designed for unattended operation. Routine, scheduled maintenance will be performed on-line, except for specified infrequent cleaning operations.

Scheduled maintenance checks shall not be required more than once every 24 hours. Scheduled maintenance may include, but not be limited to: air filter cleaning and replacement; battery cleanliness and battery voltage level checks; daily semi-automatic system tests from the SAT; lamp and meter checks; and general cleanliness requirements.

The total 24-hour normal maintenance burden for an operating system, scheduled and unscheduled, shall not average more than two man-hours (T) / one man-hour (O).

5.2 Supply

User Manuals will be provided to the operators and maintenance technicians by the equipment provider (vendor) and will include operator procedures, diagnostic testing/SAT use, and replacement procedures. No special tools or diagnostic equipment will be required for equipment replacement.

5.3 Support Equipment

Standard support equipment for the Interoperability Switch is the system administration terminal (SAT) described in paragraph 4.3.4 HMI which will handle system diagnostic testing. No special test equipment will be required to maintain or operate the unit. The vendor will provide software upgrades as needed/required and will provide software development services to the buyer for new features as requested.

5.4 Training

Training will be provided by the equipment provider to a system trainer (via a train-the-trainers session) and to the users and operators at the installed site at a time convenient to the users and operators. The training curriculum will be designed to ensure users understand and are fully capable of operating and using all features of the system. Knowledgeable staff members of the equipment provider will also be made available by phone (via a Help Desk type arrangement) should a user or operator need assistance with any part of the proposed system.

5.5 Transportation and Facilities

It is anticipated that this system will most often be used in a fixed station. If the proposed system is to be mobile or used in the field, it will be transportable via truck or van and will be able to be lifted by two or fewer personnel. Sufficient 115V power and cables will be needed to connect the communications interoperability switch to the radios and other equipment necessary to provide connectivity and interoperability commensurate with the event. Any training needed in the field can be provided as on the job training with no special facilities needed.

6 Force Structure

One Interoperability Switch system will typically be required at each emergency operating center (EOC) or similar type communications center. The proposed system will be modular and scale-able (or sizeable) to have enough capacity and interface boards necessary to interface all of the radios, integrated voice terminals, telephones and other communications devices needed by the center personnel to conduct their mission.

Additional systems can be supplied to mobile platforms (vans or trucks) if an EOC or other shore based center is not within communications range of the event.

The high reliability of the system (para. 4.3.5) dictates only a minimum amount of spares needed for interface boards, power supplies and communications devices

7 Schedule

Demonstration of an initial operational capability is required within 3 months (T) / 1 month (O) after executed SECURE Agreement. For the purpose of this effort, initial operational capability is defined as installation and field demonstration of one fully operational communications interoperability switch system to include one SAT and at least two radios, two integrated voice terminals, two telephones, and one other wireless device (such as a cell phone.)

A fully operational system will be required within 9 months (T) / 6 months (O). A fully operational system includes the communications interoperability switch with interface boards, system administration terminal (SAT), and all necessary integrated voice terminals supplied by the proposed system vendor. Radios and other communications devices (telephones, wireless systems) to interface with the communications interoperability switch are typically separate from the communications interoperability switch system and may have different lead times if they are not already available at the site.

8 System Affordability

An Individual unit price cost for such an communications interoperability switch shall cost less than \$200K (T) / \$150K (O).

9 Appendixes

List of Acronyms

CM – Class Mark

COS – Class of Service

COTS – Commercial off the Shelf Equipment

EOC – Emergency Operations Center

ISDN – Integrated Services Digital Network

KPP – Key Performance Parameter

MTBF – Mean Time Between Failures

POTS – Plain Old Telephone System

PSTN – Public Switched Telephone Network

RDT – Responsible Dial Terminal

SAT – System Administration Terminal

IVT – Integrated Voice Terminal