On telecom services and the DISN evolution

Manfred Sneps-Sneppe

***Abstract*—Communication specialists around the world are facing the same problem: shifting from circuit switching to packet switching. In 2006, the Pentagon adopted a new plan for the next 15 years entitled Joint Vision 2020. The plan announced a DISN (Defense Information System Network) paradigm shift: the transition from SS7 signaling to IP protocol. It is assumed that the IP protocol will be the only means of communication between the transport layer and applications. The article is devoted to the discussion of the telecommunications service evolution. We will provide examples to illustrate the difficulties that complicate the transition from CS to PS, to web-oriented services. A sort of birthmarks on the DISN remain the "Red Phone" network (Defense Red Switched Network) based on ISDN technology, as well as databases of AIN (Advanced Intelligent Network) and SCE (Service Creation Environment) for AIN services.**

***Keywords*— circuit switching; packet switching; SS7; intelligent network; soft-switch; DISN; SIP; AS-SIP.**

I. Introduction: the telecom paradigm shift

Communication specialists around the world are facing the same problem: shifting from circuit switching (CS) to packet switching (CS). Let us use the Pentagon's Defense Information System Network (DISN) as a case, as a bright illustration of many troubles on the road from circuit switching to packet switching, on the road to the unified services.

In 2006, the Pentagon adopted a new plan for the next 15 years entitled Joint Vision 2020. The plan announced a DISN paradigm shift: the transition from SS7 signaling to IP protocol [1]. It is assumed that the IP protocol will be the only means of communication between the transport layer and applications. However, the timing of this transition was not announced in the plan.

Underlying this internetworking convergence layer, all types of DoD-relevant physical transport media and technologies are supported (Fig. 1). For instance, this includes copper cable, optic-fiber cable, SATCOM, and tactical wireless (RF and optical). This enables a deployed tactical user to collaborate in real time (without a priori communications planning) with an intelligence analyst through mobile ad hoc networks, theater networks, SATCOM, and terrestrial fiber networks (all on a transaction-based, variable trust level).

As for today, DISN is based on circuit switching (more specifically, on the SONET standard for the optic cables functioning), and the information is coded according to the time division multiplexing (TDM) telephone standard (Fig. 2).

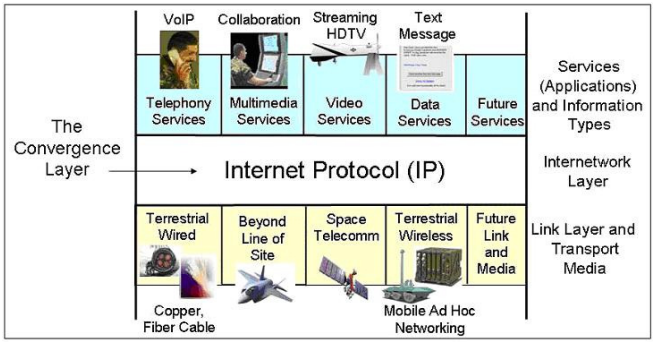

Fig. 1. DISN Internetworking Convergence Layer

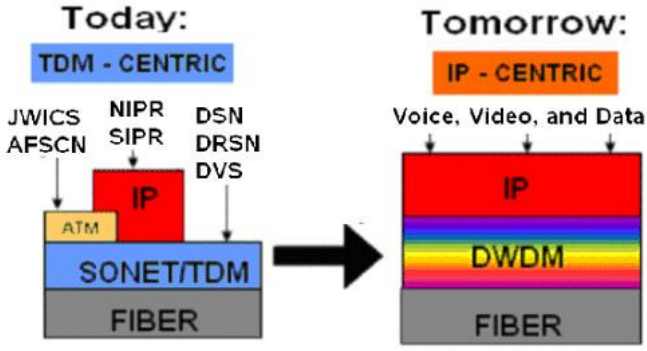

Fig.2 DISN: from circuit switching to packet switching.

This circuit switching network is currently used by the major military communication networks of the Pentagon: the Defense Switched Network (DSN) telephone network; the Defense Red Switched Network (DRSN) secure switched network; the DISN VIDEO (DVS) video conferencing network. Besides, DISN contains classified networks: JWICS (Joint Worldwide Intelligence Communications System) and AFSCN (Air Force Satellite Control Network) working in the ATM network; NIPRNet (Non-classified Internet Protocol Router Network) and SIPRNet (Secret Internet Protocol Router Network) working in the IP network, and some more. The recent revision of DISN classification: networSBU IP Data (formerly known as NIPRNet), Secret IP Data (formerly known as SIPRNet), TS/SCI IP Data (formerly known as JWICS), Multilevel Secure Voice (formerly known as DRSN) [2].

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II. On the orientation towards AIN: Joint Vision 2010

The Defense Information Systems Network (DISN) belonging to the Pentagon is the world's largest departmental network. The DISN has been developed since the early 1990s. This is a global network. It is intended to provide communication services by transmitting different types of information (voice, data, video, and multimedia) in order to perform the efficient and secure control of military, communications, intelligence, and electronic warfare media. In 1996, the state of the DISN was panned. First of all, due to the low level of integration of members of the DISN networks, significantly limiting the interaction capabilities within a single network and preventing the effective unified management of all its resources. In particular, there was noted the complexity of the interaction of stationary and field (mobile) components of the core network due to different standards being used, the types of communication channels (analog and digital), the services, and the capacity (the bandwidth of mobile components is significantly lower than that of stationary ones).

In the development of the second phase of the DISN network, the DISA agency has taken an unprecedented step for the Department of Defense (DoD): it required the usage of only the finished commercial products in the field of new information and network technologies. The emphasis was placed on open systems, which are based on national standards, and the latest commercial technologies and services available on the market (COTS, commercial-off-the-shelf).

These requirements are reflected in the 15-year program of weapons development entitled Joint Vision 2010, which the United States Joint Chiefs of Staff adopted in October 1996. Regarding the means of communication, the Advanced Intelligent Network (AIN), the highest achievement in the art of circuit switching developed by BellLabs, was chosen. Note that the breakup of the Bell System was mandated in 1982, long time before the DISA solution.

The intelligent network (IN) is an architectural concept that enables the real-time execution of network services and customer applications in a distributed environment consisting of interconnected computers and switching systems. The IN concept is a sequence of ISDN (Integrated Services Digital Network). The essence of ISDN is the integration of both analog or voice data together with digital data over the same network.

Beginning in the early 1980s, the IN was applied to the development of new services in wireline telephone networks. Many of the desirable properties of the IN architecture are based on three major principles of independence:

•Service independence (meaning that a wide variety of services can be composed using a set of common building blocks),

•Separation of basic switching functions from service and application functions, and

•Independence of applications from lower-level communication details.

The main concepts (functional view) surrounding IN services are connected with SS7 architecture:

- Service Switching Point (SSP). It is co-located with the telephone exchange itself, and acts as the trigger point for further services to be invoked during a call.
- Service Control Point (SCP). It is a separate set of platforms that receive queries from the SSP. The SCP contains service logic which implements the behavior desired by the operator, i.e., the services. During service logic processing, additional data required to process the call may be obtained from the SDF. The logic on the SCP is created using the SCE.
- Service Data Point (SDP). It is a database that contains additional subscriber data, or other data required to process a call. For example, the subscriber's prepaid credit which is remaining may be an item stored in the SDF to be queried in real time during the call.
- Service Management Point (SMP). It is a platform or cluster of platforms that operators use to monitor and manage the IN services. It contains the management database which stores the services configuration, collects the statistics and alarms, and stores the Call Data Reports and Event Data Reports.
- Service Creation Environment (SCE). This is the development environment used to create the services present on the SCP. Services can be composed using a set of common building blocks.
- Intelligent Peripheral (IP) This is a node which can connect to both the SSP and the SCP and delivers additional special resources into the call, mostly related to voice data, for example play voice announcements or collect DTMF tones from the user.

The AIN (Fig. 3) provides integrated "one stop" end user services, such as voice, data, video, e-mail, images, office applications, and 800 services. SS7 is a means by which elements of telephone networks exchange information. Information is conveyed in the form of messages. SS7 defines the procedures for the setup, ongoing management, and clearing of a call between users. The key points of AIN are the following: Service Control Point and Database of services, as well as TCAP (Transaction Capabilities Application Part) - a main protocol in the SS7 protocol stack, providing access to databases.

Intelligent Peripheral also plays an important role: its functions include tone generation, voice recognition, speech and data compression, dialing recognition, and much more, including tactical and strategic services for personnel identification. The Adjunct provides the same operation as the SCP but is configured for one or fewer services for a single switch. The Network Access Point (NAP) is a switch that has no AIN functions. It is connected off a SSP, and interfaces to trunks with SS7 messages. It will route the call to its attached SSP or AIN services based on the called and calling number received. Channel switching network subscribers, as well as packet switching network subscribers, can be AIN users. Point out the attention to the Service Creation Environment (SCE) as a standardized means for

service software development.

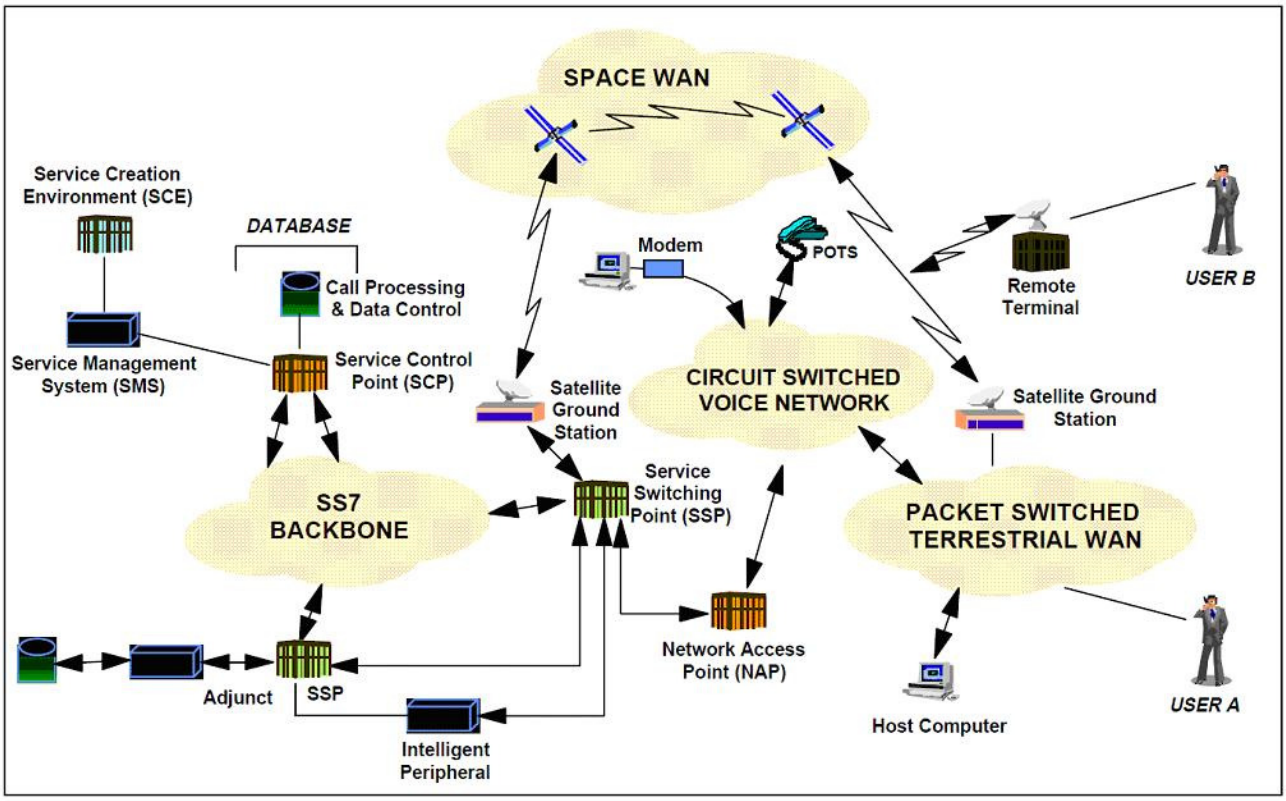

Fig. 3. Advanced Intelligent Network (AIN) Service Architecture

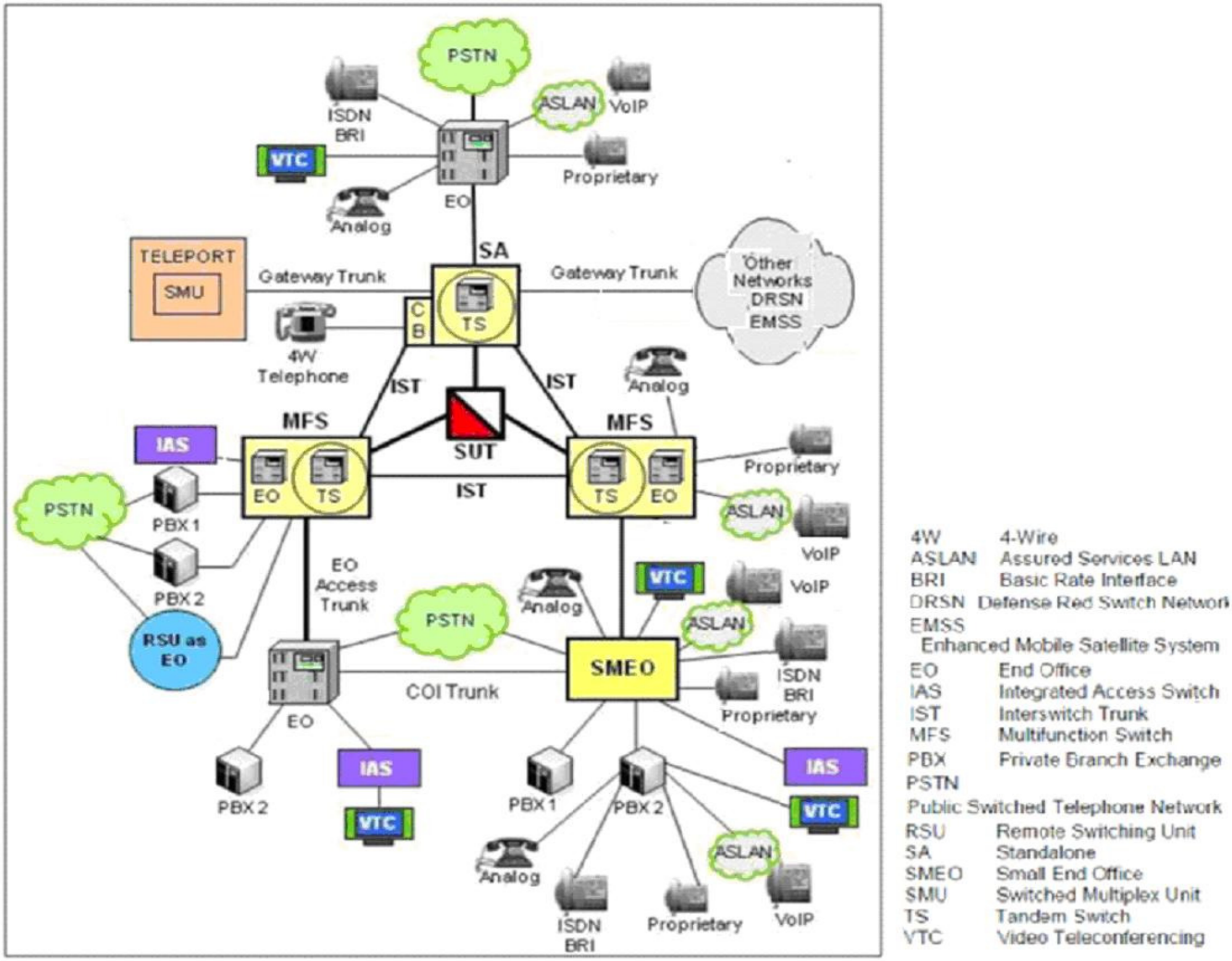

Fig. 4. DISN (Defense Information System Network): the current state [4].

Analog and ISDN BRI telephone support and many ISDN features for voice services are mandatory:

• Call Forwarding: on Busy Line, Don't Answer, Selective Call Forwarding

• Multi-Level Precedence and Preemption (MLPP): Interactions With Call Forwarding, at a Busy Station, No Reply at Called Station

• Precedence Call Waiting: Busy With Higher Precedence Call, Busy With Equal Precedence Call, Busy With Lower Precedence Call, No Answer, Line Active With a Lower Precedence Call, Call Waiting for Single Call Appearance VoIP Phones

• Call Transfer: at Different Precedence Levels, at Same Precedence Levels

• Call Hold

• Three-Way Calling .

The same is for Video Calls: Call Forwarding, Call Transfer, Call Hold, Three-Way Calling, Calling Number Delivery.

A service switching point (SSP) uses TCAP to query a signaling point control point (SCP) to determine the routing number(s) associated with a dialed 800, 888, or 900 numbers. Calling card calls are also validated using TCAP query and response messages.

The SS7 network is, figuratively speaking, the nervous system of a DISN switched network up to resent time. Figure 4 originates from the documentation on testing the SS7 network as the part of the DISN network conducted by Tekelec in 2011 [4]. The center of the diagram is occupied by the system under test (SUT) block, which is the SS7 network undergoing the test. That is, within the DISN network, the connections are established by means of SS7 signaling and, in the periphery, devices of any type are used. The devices are connected by any protocols: 4-wire (4W); classified LAN (ASLAN); ISDN BRI; Internet telephony (VoIP); video conferencing (VTC); any proprietary protocol; a link via communication satellites to remote telephone networks and tactical networks at theaters of military operations (STEP/TELEPORT).

From above an important conclusion follows: the DISN network tends to adopt new terminal equipment (to a large extent, this is IP media), but the SS7 network retains its central position till now. The presence of the SS7 network is not an obstacle to the transition to IP protocol.

III. Transition from AIN to all-IPMath

The most important step for DISN modernization is the replacing of channel switching electronic Multifunctional switches (MFS) by packet switching routers. The transition phase is based on Multifunctional SoftSwiches (MFSS).

Multifunction SoftSwitches (MFSS) Wide Area Network SoftSwitches (WAN SS) .

Figure 5 shows the reference model for MFSS [6]. The left side shows the traditional telephony protocols CCS7, ISDN PRI, and CAS used for connections with the "old" channel switching networks. MFSS interfaces the circuit-switched based external TDM network and the IP backbone network will also control the calls that are originating from the external Public Switched Telecommunications Network (PSTN)/Integrated Services Digital Network (ISDN). So, MFSS will also needs to provide ISUP-SIP inter-networking function (IWF). It is expected that TDM switching portion of the MFSS will be retired as soon as all users/systems migrate to IP.

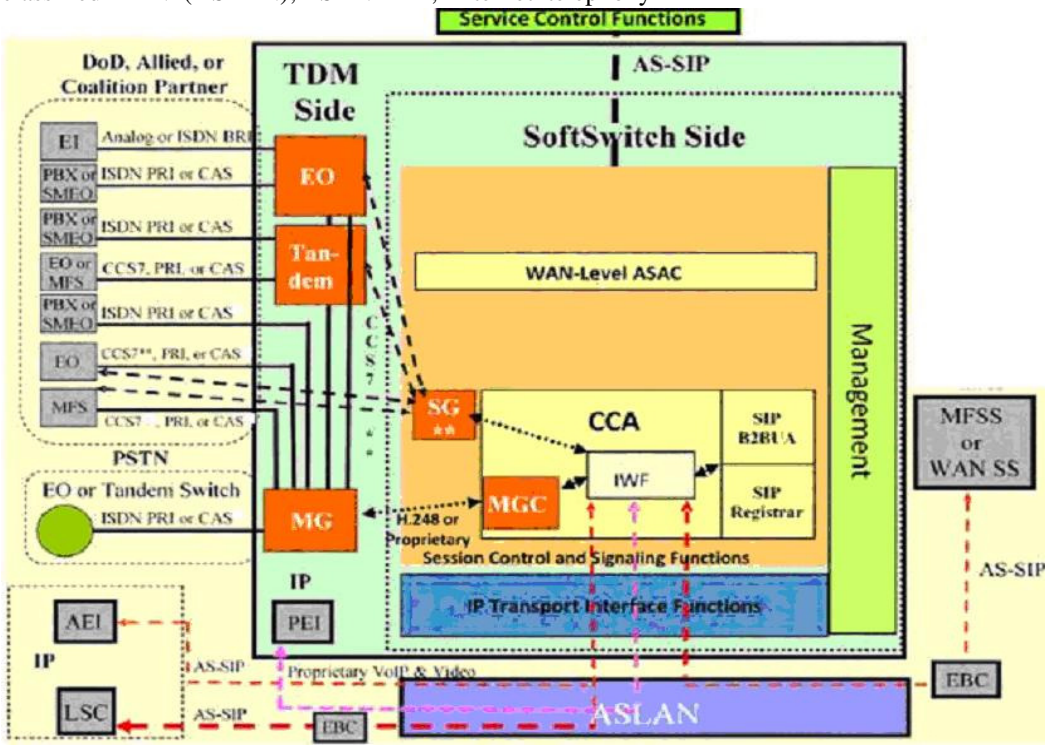

Fig. 5 Reference model for Multifunction SoftSwitch (MFSS) [6].

The MFSS provides all required PSTN/ISDN interface functions, including ISUP, CCS7/SS7, and Channel Associated Signaling (CAS) and media conversion. A signaling gateway (SG) deals with all signaling protocols such as ISUP, CCS7/SS7, and CAS. The MFSS also operates as a media gateway (MG) between TDM circuit-switching and IP packet-switching under the control of the media gateway controller (MGC) while communications control protocol like H.248 is used between MG and MGC.

IV THE TARGET DISN INFRASTRUCTURE

By connecting the satellite system to DISN, Teleports were one of main architectural joints of a grander network project called the Global Information Grid (GIG). GIG was intended to hook all US defence and intelligence systems, and surveillance and weapons platforms such as drones, into the same network using internet technology. This dedicated military internet connected more than 3,500 US military facilities and bases in 88 countries, said a DoD Task Force on Intelligence Integration in 2008 [5]. The DISN was its backbone, connecting the major bases. Teleports are linking satcoms to DISN in eight locations around the world, including two on the UK route between Stuttgart and Lemonnier - Lago Patria, Italy; and Landstuhl/Ramstein, Germany. The others – according the US Navy's 2013 Program Guide – were: Bahrain; Wahiawa, Hawaii; Fort Buckner, Okinawa, Japan; Camp Roberts, California; and Northwest, Virginia. DoD's 2014 budget cited another Teleport at Guam, a US island in the Philippines.

The Global Information Grid (GIG) Bandwidth Expansion (GIG-BE) provides a secure, optical terrestrial network that delivers very high-speed classified and unclassified IP services to key operating locations worldwide. The DoD's vision is a "color to every base." It implies that every site has an OC-192 (10 gigabits per second) of usable IP dedicated to that site.

The GIG-BE program is one costly but rather simple step towards the DISN migration to IP end-to-end.

The target DISN infrastructure contains two level switching nodes: Tier0 and Tier1. Tier0 geographic cluster typically consists of at least three Tier0 SoftSwitches (SSs). As the distance between the clustered SSs must be planned so that the RTT does not exceed 40 ms and propagation delay equals 6 μs/km thus distance between Tier0 should not exceed 1860 miles. The classified signaling environment is unique in that it will use a mix of existing vendor-based H.323 and AS-SIP signaling during the transition period to all DISN CVVoIP. In addition, a unique MG capability exists as part of a Tier0 SS.

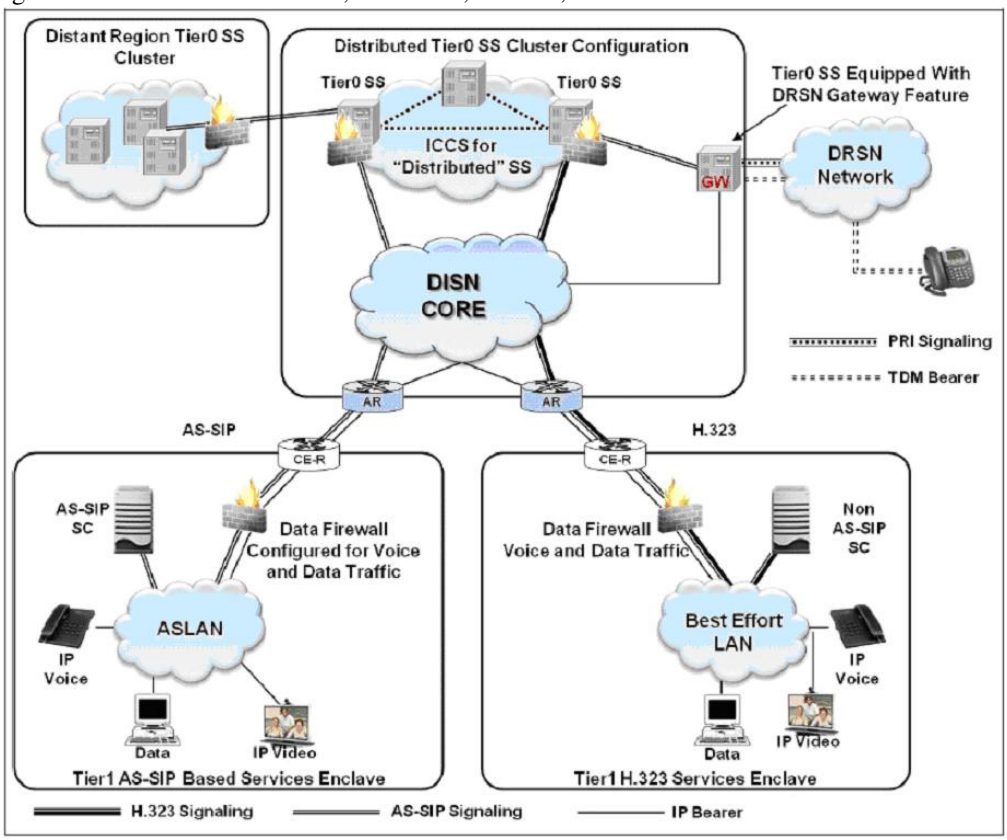

Fig. 6. DISN Classified VoIP and Video (CVVoIP) Signaling Design [3].

The classified Voice and Video Signaling Design is shown in Figure 3. Currently, the classified voice and video services employ H.323, and will migrate to AS-SIP signaling in the future. Duration migration, both H.323 and AS-SIP signaling will be employed in classified VVoIP. Classified VVoIP interfaces to the TDM Defense RED Switch Network (DRSN) via a proprietary PRI. The Common Channel Signaling 7 (CCS7) network is being phased out and replaced by PRI trunks. The TDM End Offices (EOs) use PRI for signaling to the TDM switching part of the SS.

The SSs use AS-SIP between themselves to set up IP-to-IP EI sessions across the DISN IP WAN.

During a transition period, H.323 and AS-SIP will coexist at certain locations. Thus, CAS and PRI in the DRSN has to interoperate with H.323 signaling in the VoSIP Pilot to be followed by H.323 and AS-SIP interoperating in the CVVoIP system until all IP services are via AS-SIP.

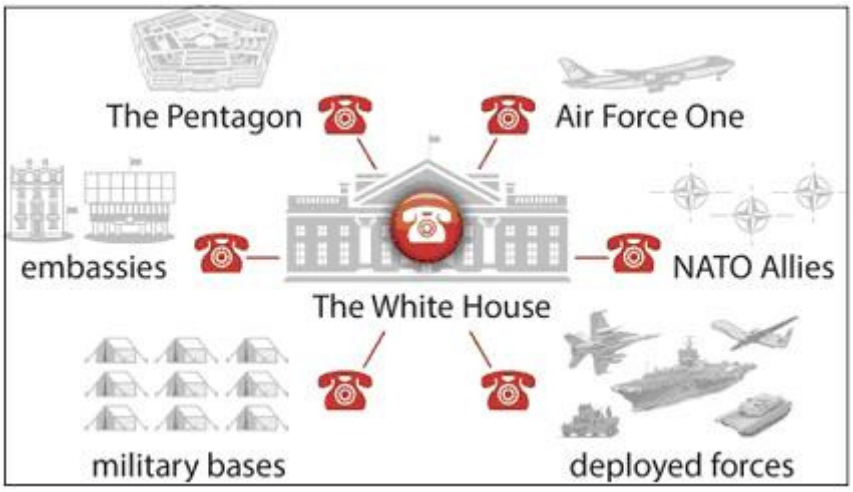

Fig. 7. Secure Terminal Equipment, STE; note slot in front for Crypto PC Card (left). The DRSN architecture (right).

ISDN technology oriented the Defense Red Switch Network (Red phone network) is some kind of birthmark in the environment of AS-SIP. DRSN is a dedicated telephone network which provides global secure communication services for the command and control structure of the United States Armed Forces (Fig. 7). The network is maintained by the Defense Information Systems Agency (DISA) and is secured for communications up to the level of Top Secret SCI. Secure Terminal Equipment (STE) is designed to use ISDN telephone lines 128 Kb/s.

The origin of DRSN is going back to 50-years old history – to Moscow–Washington hotline that allows direct communication between the leaders of the United States and Russia. This hotline was established in 1963 and links the Pentagon with the Kremlin.

V Comments on MFSS Service Control Function

Unified Capabilities provides the ability to seamlessly integrate voice, video, and data applications services so they are delivered ubiquitously across a secure and highly available single protocol network infrastructure EoIP environment due the means of Service Control Function [6]. It comprises of 19 servers (Fig. 8) and the announced list of services is a kind of museum: there are all services starting from the ISDN era up to nowadays web services [8].

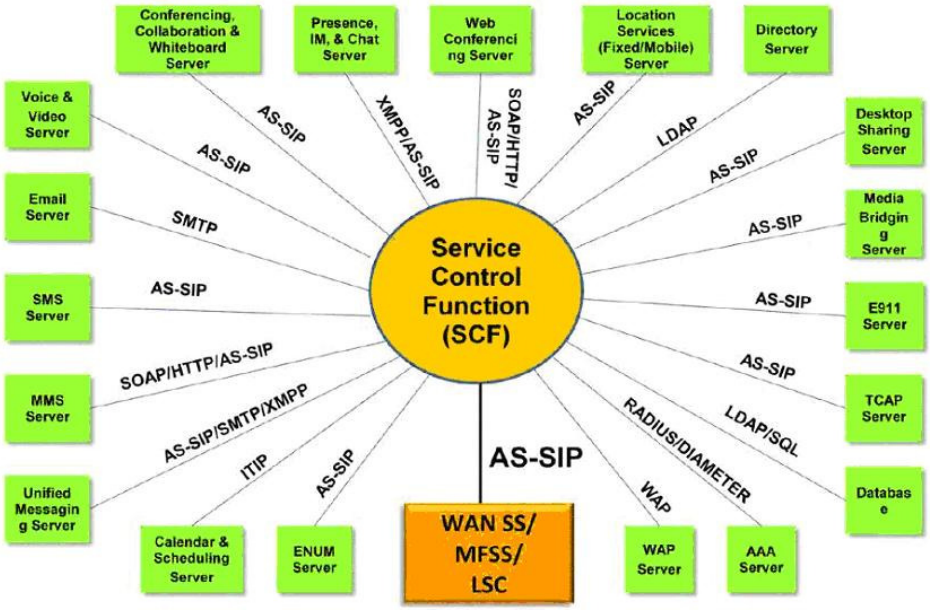

Fig. 8. The architecture of Service Control Function [6, 7]. Protocols and services: AS-SIP = Assured Service – Session Initiation Protocol, SOAP = Simple Object Access Protocol, HTTP = HyperText Transport Protocol, LDAP = Lightweight Directory Access Protocol, SQL = Structured Query Language, RADIUS = Remote Authentication Dial In User Service, DIAMETER = an enhanced version of RADIUS, WAP = Wireless Access Protocol, ITIP = iCalendar Transport Independent Interoperability Protocol, SMTP = Simple Mail Transfer Protocol, AAA = Authentication, Authorization, and Accounting,

TCAP = Transaction Capabilities Application Part, ENUM =E.164 Number, IM = Instant Messaging, MMS = Multimedia Messaging Service, SMS = Short Message Service

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- Call Forwarding: on Busy Line, Don't Answer, Selective Call Forwarding
- Multi-Level Precedence and Preemption (MLPP): Interactions With Call Forwarding, at a Busy Station, No Reply at Called Station
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- Call Transfer: at Different Precedence Levels, at Same Precedence Levels
- Call Hold
- Three-Way Calling.

The same is for Video Calls: Call Forwarding, Call Transfer, Call Hold, Three-Way Calling, Calling Number Delivery. Besides, there are several unsolved problems, name some of them.

E.911 (Emergency Call) Application Server. The Army UC Emergency (E-911) call architecture is a hybrid architecture that leverages the capabilities of both IP and TDM network because the E-911 call architecture using AS-SIP call control protocol is still emerging. Public Safety Access Points (PSAPs)-based E-911 call architecture over the TDM network is quite stable and is serving the needs for a long time. The seamless integration of the E-911 architecture over both IP and TDM network has been termed as the Next Generation 911 (NG-911) architecture. While a great deal of progress has been made, Next Generation 911 (NG-911) standards are still a work-in-progress. The NG-911 infrastructure (e.g., Emergency Call Routing application servers and associated databases) is expected to take several years to implement.

Transaction Capabilities Application Part (TCAP) Application Server. The TCAP enables the deployment of advanced intelligent network services by supporting non-circuit related information exchange between signaling points using the Signaling Connection Control Part (SCCP) connectionless service in common signaling system 7 (SS7) networks of the TDM network. However, recent development of stream control transmission protocol (SCTP) that also offer connectionless services has made the use of the TCAP applications versatile for being used equally over both IP and TDM network replacing the SCCP. A service switching point (SSP) uses TCAP to query a signaling point control point (SCP) to determine the routing number(s) associated with a dialed 800, 888, or 900 numbers. Calling card calls are also validated using TCAP query and response messages. In fact, the TCAP IP Gateway (TIGW) is used in the AS-SIP-based Army UC network for intelligent TDM/Wireless routing as well as being the legacy signaling gateway for application servers for seamless services over both IP and TDM network.

The detail of the TCAP services architecture will be addressed in the future.

Short Message Service (SMS) Application Server. The SMS server is of special interest for military applications. SMS allows the exchanges of short messages between fixed landline, satellite link, or mobile phone devices. It has become an integral part of the service component cellular mobile phones like voice and web services. This service is also offered to the Army UC mobile users connected to the Army LWN via external cellular carriers' networks. SMS can also be used for emergency services, invoking voice calls, or many other services. However, AS-SIP protocol is used for communications with the SCF. In the future, the detail architecture the SMS application will be described in detail, especially for emergency services.

Location Services Application Server. The Location Service is an application that manages the location information of the user that uses UC services over the Army worldwide global LWN war-fighter networks. In static war-fighter networking environments, a terminal's network address serves two purposes: End-point identifier and Location identifier. That is, a single network address servers these two purposes simultaneously. In the AS-SIP network, the location management AS keeps the SIP contacts and other information in the database. The SIP registrar stores contacts and other information of users in the location management server. An MFSS or LSC communicates with the location management server for address resolutions in order to route SIP messages to users or other servers such as MFSSs or LSCs. This location management AS should also have distributed architecture. However, it is expected that the communication protocol between the location management AS and the MFSS or LSC will be AS-SIP. The key aspect of the Army UC war-fighter network is mobility where war-fighters will be moving from one place to another and the wireless communications will be predominant especially for the dismounted soldiers. Location management involves maintaining location information as mobiles power-on, move or power-off. The important point is that mobility prevents using a single address for both end-point identifier and location identifier purposes. Both end-point identifier and location identifier are needed, and the location management application needs to keep mapping between an end-point identifier and its location identifier turning into basically a directory lookup problem.

Two primitive operations are done by the location management server: Lookup operation and Update operation. The lookup operation is the search, find, paging, and/or locating procedure by which the war-fighter network finds the location of the mobile. It is required when an AS-SIP call (message) is placed (to be delivered) to a user. The update operation is the tracking, move, and/or registration procedure by which the network elements update information about the location of the mobile. It is required when a user changes its "location." The information

gathered during updating/tracking is used during the locating operation. The AS-SIP has the inherent capability to manage the user mobility, terminal mobility, and service/session mobility, as described earlier, at the applications layer. The impact of mobility in the lower network (e.g. mobile IP) and link/physical layer (e.g. MAC protocols/modulation schemes in wireless mobile environments) will be addressed accordingly. In the future, the detail architecture the SMS application will be described in detail.

VI COMMENTS ON U.S. ARMY UNIFIED CAPABILITIES

According to the up-to date tendencies, the future services should be web-oriented (Table 1).

Table 1. Service descriptions

Service	Description
Email and Calendaring	Provides for users to send messages to one or many recipients with features such as priority marking, reports on delivery status and delivery receipts, digital signatures, and encryption. Calendaring allows the scheduling of appointments with one or many desired attendees.
Instant Messaging and Chat	The capability for users to exchange one-to-one ad hoc text messages over a network in real time. Instant Messaging is not the same as and must not be confused with signaling or equipment messaging; IM is always user generated and user initiated. Chat provides the capability for two or more users operating on different computers to exchange text messages in real time. Chat is distinguished from IM by being focused on group chat or room-based chat. Typically, room persistence is a key feature of multi-user chat, in contrast with typically ad hoc IM capabilities.
Rich Presence	Allows contact to be achieved with individuals based on their availability as displayed by presence information from multiple sources, including IM, telephone, and mobile devices.
Unified Messaging	Provides access to voicemail via e-mail or access to e-mail via voicemail
Video Conferencing	Provides multiple video users with the ability to conduct video and voice collaboration with a variety of room controls for displays of the participants often with a variety of scheduling tools.
Voice and Video (Point-to-Point)	Provides two voice and/or video users with the ability to be connected End-to-End with services that can include capabilities such as voicemail, call forwarding, call transfer, call waiting, operator assistance, and local directory services
Voice Conferencing	Provides multiple voice users with the ability to conduct a collaboration session.
Web Conferencing and Web Collaboration	Provides for multiple users to collaborate with voice, video, and data services simultaneously using web page type displays and features.

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