

Application Note for V.150.1 (Modem over IP) Interoperability

May 21, 2018
Issue 1.4

Abstract:

This document is designed to aid in the basic knowledge of modem modulation modes and an explanation of the V.150.1 Modem-over-IP protocol. This knowledge should be understood for interoperability testing and troubleshooting enterprise solutions involving both US Government (DoD and Non-DoD) SIP networks and commercial telemetry applications.

This document is not a substitute for the ITU-T specifications of V.150.1 (Modem-over-IP Networks). It is meant to enable one to have a good operational understanding of the terminology, concepts and the administrative screens for implementing an enterprise solution to support this protocol.

There are two primary applications:

- US Government (DoD and Non-DoD) SIP networks which deploy secure communication between secure station sets and gateways.
- Commercial applications which use modem-like communication for telemetry applications

This document will be updated on a regular basis as information about new vendor products becomes available and interoperability testing is successfully conducted.

This document has been updated to include both the G430 and the G450 media gateways. Many of the diagrams are shown with G450, but these examples will operate correctly with both gateway products.

Table of Contents:

1	Reference Documentation	4
2	Document Change History.....	6
3	Terminology and Acronyms.....	6
4	Market Introduction	10
4.1	Key Components for CM/GW Solution.....	10
5	Applications.....	11
5.1	DoD Application	11
5.2	Commercial Application	13
6	Technology Summary.....	15
6.1	Gateway Protocol Architecture for Modem-over-IP Application.....	15
6.2	CM to Gateway Media Services Capabilities Exchange.....	16
6.3	CM to Gateway Provisioning & Signaling Flow Sequence.....	17
7	DSP Hardware Overview	18
7.1	DSP Card Support for the G450 Gateway	18
7.2	DSP Card Support for the G430 Gateway	19
7.3	DSP Point Cost Characteristics	19
7.3.1	MP 160 DSP Point Cost Characteristics.....	19
7.3.2	MP 120 DSP Point Cost Characteristics.....	20
7.3.3	MP 80 DSP Point Cost Characteristics.....	20
7.4	Media Gateway Channel Allocation for the G450 Gateway.....	20
7.5	DSP Channel Usage Discussion for the G450 Gateway.....	21
8	End to End Flow Diagrams for V-Series Modulation Exchange	23
8.1	Establishing Modem Relay with V.32 Modem.....	24
8.2	Establishing Modem Relay with V.34 Modem.....	25
8.3	Establishing Modem Relay with V.90 Modem.....	27
8.4	Establishing Modem Relay with V.92 Modem.....	28
9	List Measurement SAT Performance Tool.....	29
10	List Trace SAT Diagnostic Tool	30
10.1	Verbose Mode for List Trace.....	31
11	Types of Secure Phone Devices	32
11.1	General Dynamics	32
11.1.1	Sectera® vIPer™ and TalkSECURE™ vIPer™ Universal Secure Phone.....	32
11.1.2	Sectera® Wireline and TalkSECURE™ Wireline Terminal.....	33
12	Fax Operation in a CM-6.2+ V.150.1 (MoIP) Environment	34
12.1	V.8 Fax Calling, V.8 Fax Answer	36
12.2	Low-Speed Fax Calling , V.8 Answer.....	38
12.3	Low Speed Fax Answer.....	39
13	Types of V.150 Compatible Gateways	40
13.1	AudioCodes	40
13.1.1	Mediant 3K	40
13.2	NET Gateway	41
14	Types of Commercial Data Modems.....	41
15	Appendix A: Administration of Communication Manager	42
15.1	Communication Manager Administration Settings for ‘ip codec set’	42
15.2	RTP Payload Type Assignment.....	46
16	Appendix B: Administration of General Dynamics Sectera vIPer Phones	47

16.1	General Dynamics vIPer Administration Settings.....	47
16.1.1	System Information.....	47
16.1.2	Phone Settings.....	47
16.1.3	Network Settings.....	48
16.1.4	SCCP Configuration	48
16.1.5	SIP General Settings	48
16.1.6	SIP Feature Settings.....	51
16.1.7	SIP Dial Settings	52
16.1.8	SIP Security Settings.....	53
16.1.9	SIP Codec Settings.....	53
16.1.10	V.150 Parameters	54
16.1.11	Code Upgrade.....	55
16.1.12	Change Password	55
16.1.13	Restore Factory Defaults.....	55
17	Appendix C: Administration of General Dynamics Sectera Wireline Terminal.....	56
17.1	General Dynamics Wireline Terminal Administration Settings.....	56
18	Appendix D: Administration of NET VX1800 Gateway	57
18.1	NET VX1800 Gateway Administration Settings	57
19	Appendix E: Administration of AudioCodes Mediant 3000 Gateway	60
19.1	AudioCodes Mediant 3000 Gateway Administration Settings.....	60

1 Reference Documentation

The following ITU-T Recommendations and other references provide a complete background on the aspects of communication network setup and transmission of facsimile information.

- ITU-T Recommendation **G.711** (1988), *Pulse Code Modulation (PCM) of Voice Frequencies*
- ITU-T Recommendation **G.722.1** (2005), *Low Complexity Coding at 24 and 32 Kbits/s for Hands-Free Operation in Systems with Low Frame Loss.*
- ITU-T Recommendation **G.722.2** (2003), *Wideband Coding of Speech at around 16 Kbits/s Using Adaptive Multi-Rate Wideband (AMR-WB).*
- ITU-T Recommendation **G.723.1** (2006), *Dual Rate Speech Coding for MultiMedia Communication Transmitting at 5.3 and 6.3 Kbit/s.*
- ITU-T Recommendation **G.726** (1990), *40, 32, 24, 16 Kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*
- ITU-T Recommendation **G.729** (1996), *Coding of Speech at 8 kbit/s Using Conjugate-structure Algebraic-Code-Excited Linear Prediction (CS-ACELP)*
- ITU-T Recommendation **H.245** (2005), *Control Protocol for MultiMedia Communications*
- ITU-T Recommendation **H.248.1v2** (2002), *Gateway Control Protocol: Version 2*
- ITU-T Recommendation **H.323** (2003), *Packet-Based Multimedia Communications System*
- ITU-T Recommendation **T.38** (2004), *Procedures for Real-Time Group 3 Facsimile Communications over IP Networks*
- ITU-T Recommendation **V.8** (2000), *Procedures for Starting Sessions of Data Transmission over the Public Switched Telephone Network*
- ITU-T Recommendation **V.17** (1991), *A Two-Wire Modem for Facsimile Applications with Rates up to 14,4000 bit/s.*
- ITU-T Recommendation **V.21** (1988), *300 bits per second Duplex Modem Standardized for Use in the General Switched Telephone Network*
- ITU-T Recommendation **V.25** (1996), *Automatic Answering Equipment and General Procedures for Automatic Calling Equipment on the General Switched Telephone Network Including Procedures for Disabling of Echo Control Devices for both Manually and Automatically Established Calls.*
- ITU-T Recommendation **V.29** (1988), *9600 Bits per Second Modem Standardized for Use on Point-to-Point 4-Wire Leased Telephone Type Circuits.*
- ITU-T Recommendation **V.32** (1993), *A Family of 2-wire, Duplex Modems Operating at Data Signalling Rates of up to 9600 bit/s for Use on the General Switched Telephone Network and on Leased Telephone-Type Circuits*
- ITU-T Recommendation **V.34** (1998), *A Modem Operating at Data Signalling Rates of up to 33 600 bit/s for Use on the General Switched Telephone Network and on Leased Point-to-Point 2-Wire Telephone-Type Circuits*
- ITU-T Recommendation **V.42** (2002), *Error-Correcting Procedures for DCEs Using Asynchronous-to-Synchronous Conversion*
- ITU-T Recommendation **V.90** (1998), *A Digital Modem and Analogue Modem Pair for Use on the Public Switched Telephone Network (PSTN) at Data*

Signaling Rates of up to 56 000 bit/s Downstream and up to 33 600 bit/s Upstream

- ITU-T Recommendation **V.92**, (2000), *Simultaneous Transmission of Data and other Signals Enhancement to Recommendation V.90*
- ITU-T Recommendation **V.150.1** (2003), *Modem-over-IP Networks: Procedures for the End-to-End Connection of V-Series DCEs*
- ITU-T, *Issues List for V.150.x (V.MoIP)*, ITU-T Study Group 16, Rapporteur Q14/16, Geneva, January 27-February 6, 2009.
- ITU-T Recommendation **V.150.1 Amendment 2**, *Modem-Over IP Networks: Procedures for the End-to-End Connection of V-Series DCEs- Amendment 2: ToIP and new SPRT Data Types Support*, May 2006
- IETF RFC 2198 (1997), *RTP Payload for Redundant Audio Data*
- IETF RFC 2327 (1998), *SDP: Session Description Protocol*
- IETF RFC 2543 (1999), *SIP: Session Initiation Protocol*
- IETF RFC 2733 (1999), *An RTP Payload Format for Generic Forward Error Correction*
- IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
- IETF RFC 3264 (2002), *An Offer/Answer Model with the Session Description Protocol*
- IETF RFC 3388 (2002), *Grouping of Media Lines in the Session Description Protocol*
- IETF RFC 3389 (2002), *Real-Time Transport Protocol (RTP) Payload for Comfort Noise (CN)*
- IETF RFC 3407 (2000), *Session Description Protocol (SDP) Simple Capability Declaration*
- IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*
- IETF RFC 3711 (2004), *The Secure Real-Time Transport Protocol (SRTP)*
- IETF RFC 4566 (2006), *SDP: Session Description Protocol*
- **SCIP-215**, *US Secure Communication Interoperability Protocol (SCIP) Implementation Standard & Minimum Essential Requirements (MER) Publication*, Revision 2.2.
- **SCIP-216**, *Minimal Essential Requirements (MER) for V.150.1 Gateways Publication*, Revision 2.2.
- United States Department of Defense, *Unified Capabilities Requirements 2008*, Dec 22, 2008
- United States Department of Defense, *Unified Capabilities Requirements 2008 Change 2*, December, 2010.

2 Document Change History

EVENT	DOCUMENT DATE	CHANGE DESCRIPTION
Issue 1.0 issued for review	26 June 2013	This introduced the G450 gateway, along with its MP160 DSP module.
Issue 1.1 issued for review	12 July 2013	
Issue 1.1 issued for external release	09 Oct 2015	Stored as 162360.
Issue 1.2 issued for external release	28 March 2016	The G430 gateway, along with its MP120 DSP module, has been incorporated into this description.
Issue 1.3 issued to include more field support notes	23 April 2018	G430/G450 gateway DSP code removed limit test for round-trip delay. This enhancement helps when carrier networks introduce lots of packetization delay.
Issued 1.4 issued to remove an inaccurate note with respect to the Session Manager support of the MLPP feature.	21 May 2018	In Appendix B (Administration of vIPer phones), remove the mini section 16.1.7 (entitled “SIP MLPP Settings”). This note about Session Manager not supporting MLPP is no longer accurate. With the introduction of Session Manager R7.1, MLPP is now supported with SIP end devices.

Table 1: Document Change History

3 Terminology and Acronyms

TERM	MEANING
ANS	2100 Hz modem answer tone (defined by ITU Recommendation V.25)
ANSam	ANS with 15 Hz amplitude modulation (defined by ITU Recommendation V.8)
AS-SIP	Assured Services SIP
CD	Call Discrimination
Clear Channel	Avaya’s proprietary version of the “Clearmode” protocol
Clearmode	Bit-transparent transmission of ISDN B channels over IP. This is specified in IETF RFC 4040
CLI	Command Line Interface
CM	Avaya Aura Communication Manager
CM	Calling Menu (defined in ITU Recommendation V.8).

TERM	MEANING
COMSEC	COMmunications SECurity
CSS	Center Stage Switch (TDM inter-gateway fiber connectivity - retired)
DoD	United States Department of Defense
ECM	Error Correction Mode
FoIP	Fax over IP This refers to both Relay and Pass-Through mode.
H.248	This is an ITU-T (International Telecommunications Union – Telephony Sector) standard that specifies how a media gateway platform should communication is a media gateway controller. CM can be considered as an H.248 media gateway controller.
H.323	This is an ITU-T (International Telecommunications Union – Telephony Sector) standard that describes the signaling required between IP endpoints. This standard is largely based upon the traditional ISDN Layer 3 standard known as Q.931; however, introduces new concepts like a gatekeeper and registration.
IETF	Internet Engineering Task Force – an Adhoc standards group that works on the specification of protocols for the Internet.
IGC	Inter-gateway connection (proprietary between Avaya gateways)
IP-TLP	IP Transport Layer Protocol (i.e., SPRT)
ISDN	Integrated Services Digital Network
JITC	Joint Interoperability Test Command (US Government)
JM	Joint Menu (ITU Recommendation V.8)
MER	Minimum Essential Requirements
MG	Media Gateway – refers to a product that aggregates and translates between various types of media sources for example analog, DCP, and IP.
MGC	Media Gateway Controller
MGP	Media Gateway Processor
MM	Modular Messaging – Avaya’s older messaging product.

TERM	MEANING
MoIP	Modem over IP
MP160	New 160-channel DSP card for the G450 gateway
MP120	New 120-channel DSP card for the G430 gateway. On older G430 gateways, when the MP120 is installed, the on-board 20 channel DSP chip is disabled. Newer models of the G430 have no on-board 20 channel DSP chip. Therefore for all G430 gateways which employ a MP120 module, the DSP capacity is the same.
NVRAM	Non-Volatile Random Access Memory
PKI	Public Key Infrastructure
PSTN	Public-Switched Telephone Network – A voice telephone network offered by a traditional public service provider.
RTP	Real-Time Transport Protocol
SBU	Sensitive But Unclassified
SCIP	Secure Communications Interoperability Protocol
SCIP-215	Requirements document for MER compliant endpoints
SCIP-216	Requirements document for MER compliant gateways
SIP	Session Initiation Protocol
SM	Avaya's Session Manager – This product is responsible for routing of SIP traffic in an enterprise network.
SP	Service Provider
SPRT	Simple Packet Relay Transport (V.150.1 – Annex B)
SRTP	Secure RTP
SSE	State Signaling Events (V.150.1 – Annex C)
STE	Secure Terminal Equipment manufactured by L-3 Communications
UCR	Unified Communications Requirements
V.8	ITU-T recommendation for Inter-modem signaling used by V.34, V.90, V.92 modulation protocols
V.17	ITU-T recommendation for 2-wire modem for facsimile application up to 14,400 bps.
V.21	ITU-T recommendation for Low speed modulation used by V.8 for modem-to-modem signaling
V.25	ITU-T recommendation for modem-to-modem toned-based calling procedures that predate V.8.
V.29	ITU-T recommendation for 9600 bps modem for 4-wire circuits
V.32	ITU-T recommendation for modem modulation capable of

TERM	MEANING
	speeds up to 9600 bps
V.34	ITU-T recommendation for modem modulation capable of speeds up to 33.6 Kbps
V.42	Modem-to-modem HDLC-based error correction protocol
V.90	ITU-T recommendation for modem modulation capable of 56Kpbs (downstream) and 28Kbps (upstream)
V.92	ITU-T recommendation for modem modulation capable of 56Kbps (downstream) and 48Kbps (upstream)
V.150.1	ITU-T recommendation for Modem-over-IP Networks: Procedures for the End-to-End Connection of V-Series DCEs
VBD	Voice Band Data (synonymous with “pass-through”)
vIPer	SCIP endpoint manufactured by General Dynamics

Table 2: Terms and Acronyms

4 Market Introduction

The market introduction was as follows:

1. Delivery in May, 2013 for the Non-DoD government market with G450 gateway.
 - a. Secure protocol transported over V.150.1 modem relay
 - b. Support of secure voice terminals with V.150.1 modem relay
 - i. General Dynamics Sectera vIPer phones
 - ii. General Dynamics Sectera Wireline Terminal
 - c. Support of 3rd party gateways which support V.150.1 modem relay
 - i. NET VX gateways
 - ii. AudioCodes M3K gateway
2. Delivery in May, 2013 for the commercial market with G450 gateway.
 - a. Support of 3rd party gateways which support V.150.1 modem relay
 - b. Support of telemetry (modem over IP) applications
 - i. Data and voice traffic can be converged over a single SIP-based network.
3. Delivery in May, 2014 (CM R6.3.5) for Non-DoD government market and commercial market with G430 gateway.
4. Certified in August, 2014 with CM R6.3.6 (6.3-FP4) with full JITC testing, including the V.150.1 feature. G430/G450 now approved for all markets for the V.150.1 feature.

4.1 Key Components for CM/GW Solution

The following components are required to provide a solution for V.150.1 service:

1. **CM Software**
 - a. Requires R6.3.4 or higher.
2. **GW Firmware**
 - a. Requires Gateway load 36.25(or greater) for G430/G450 for Avaya Aura[™] R6.3.17.
 - b. Requires Gateway load 39.12(or greater) for G430/G450 for Avaya Aura[™] R7.1.3.
3. **GW Hardware**
 - a. Support of G450 requires MP160 DSP card [1 or 2 cards].
 - b. Support of G430 requires MP120 DSP card [1 card].

5 Applications

5.1 DoD Application

Figure 1 provides an overview topology for the discussion regarding government applications of modem-over-IP transport.

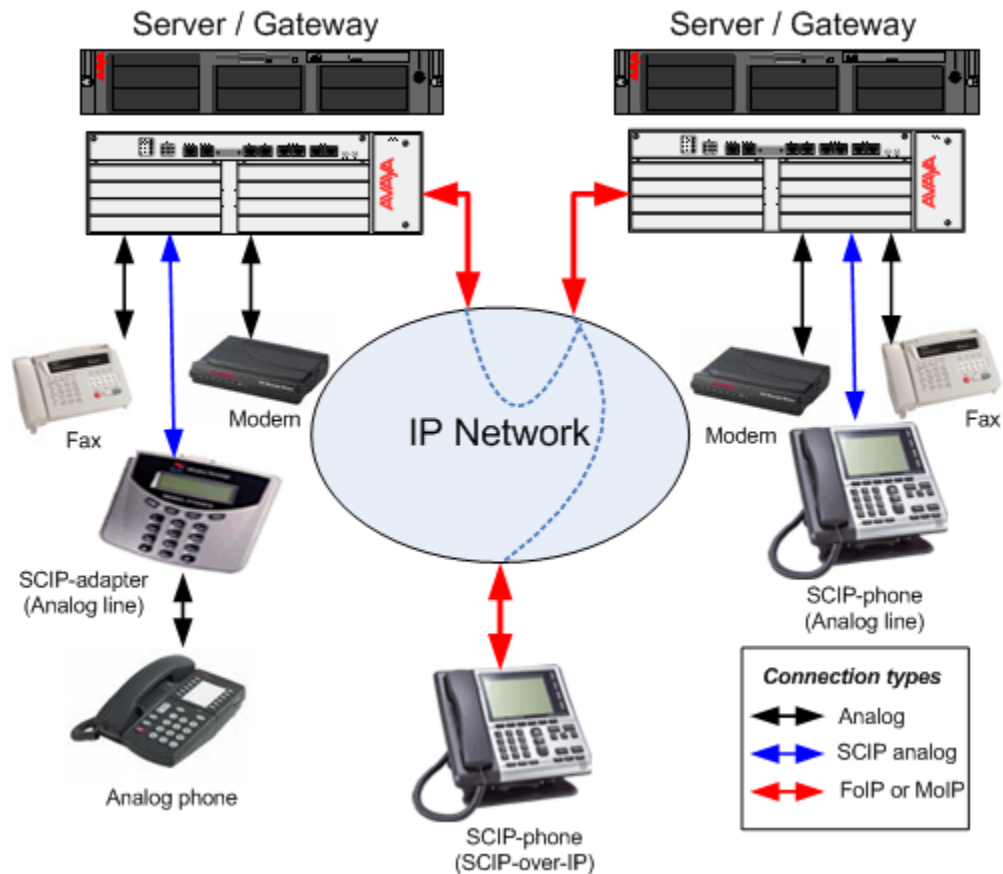


Figure 1: V.150.1 Modem Relay Network Overview

UCR 2013 requires that all media gateways support ITU-T V.150.1 Modem Relay, a need which is driven by the evolution of end-end secure voice over IP bearer. In the past, secure telephones provided end-end security by imposing encrypted digitized voice on modem (STU-III, FNBDT/SCIP) or digital (STE) carriers. A new generation of SCIP-over-IP telephones (vIPer, VoIP STE) eliminates the carrier layer, instead using IP relay techniques between peer endpoints and with endpoint-gateway connections.

The gateway's job is to provide an interworking bridge between modem relay native to SCIP-over-IP endpoints, and standardized ITU-T V.32 and V.34; enabling interoperable SCIP media transit over IP, TDM and legacy PSTN connections.

The gateway must be prepared to support a transition from audio to V.150.1 modem relay on any VoIP call having the potential for SCIP-over-IP behavior. Since a connection's potential for SCIP-over-IP behavior is not known in advance, all calls transiting the gateway must offer V.150.1 service.

UCR 2013 also requires V.150.1 Modem Relay in support of data modem traffic using V.90 and V.92 modulations.

The UCR specifies that connections incapable of negotiating modem relay for V.32, V.34, V.90 and V.92 must fall back to a G.711 transport with echo cancellation and silence suppression conditionally disabled. This rather-vague description of Modem Relay fallback sidesteps all mention of V.152, which is the V.150.1-sanctioned fallback to a VBD carrier. Similar treatment is described for fax pass-through, though that treatment is allowed to be explicitly administered by the UCR.

Figure 2 illustrates an off-site secure device wanting to employ secure data applications, along with the secure voice connection to be connected via a V.150.1 compliant G450/G430 gateway and then transit over the customer's enterprise SIP network to a SCIP-over-IP phone, which supports this same secure data application.

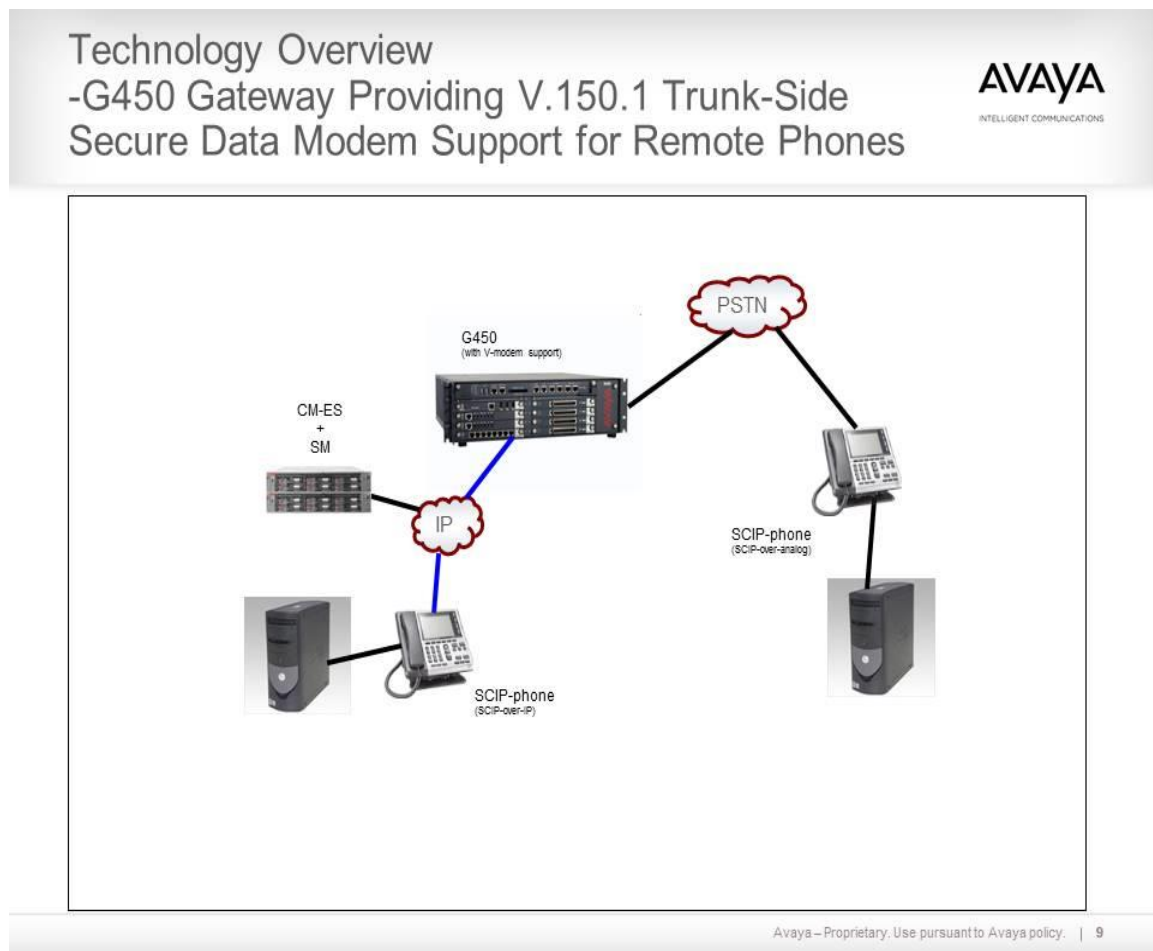


Figure 2: G450 Gateway Providing V150.1 Trunk-side Secure Data Modem Support

The CM **'ip-codec-set'** administrative form supports the specification of the features for modem, fax and clear channel. Since CM software performs trunk signaling on behalf of the gateway, it must offer these features through signaling at call setup time. In turn, CM software must send negotiated options down to the G450/G430 using H.248 messaging. The Media Gateway firmware selects only DSPs capable of supporting these features to carry the call.

5.2 Commercial Application

Telemetry applications have been around for decades, but the convergence of telemetry application transport over IP-based networks can provide large cost savings.

Fourteen areas of application usage are defined by standards groups. Some application areas that have been touched by Avaya gateway customers are:

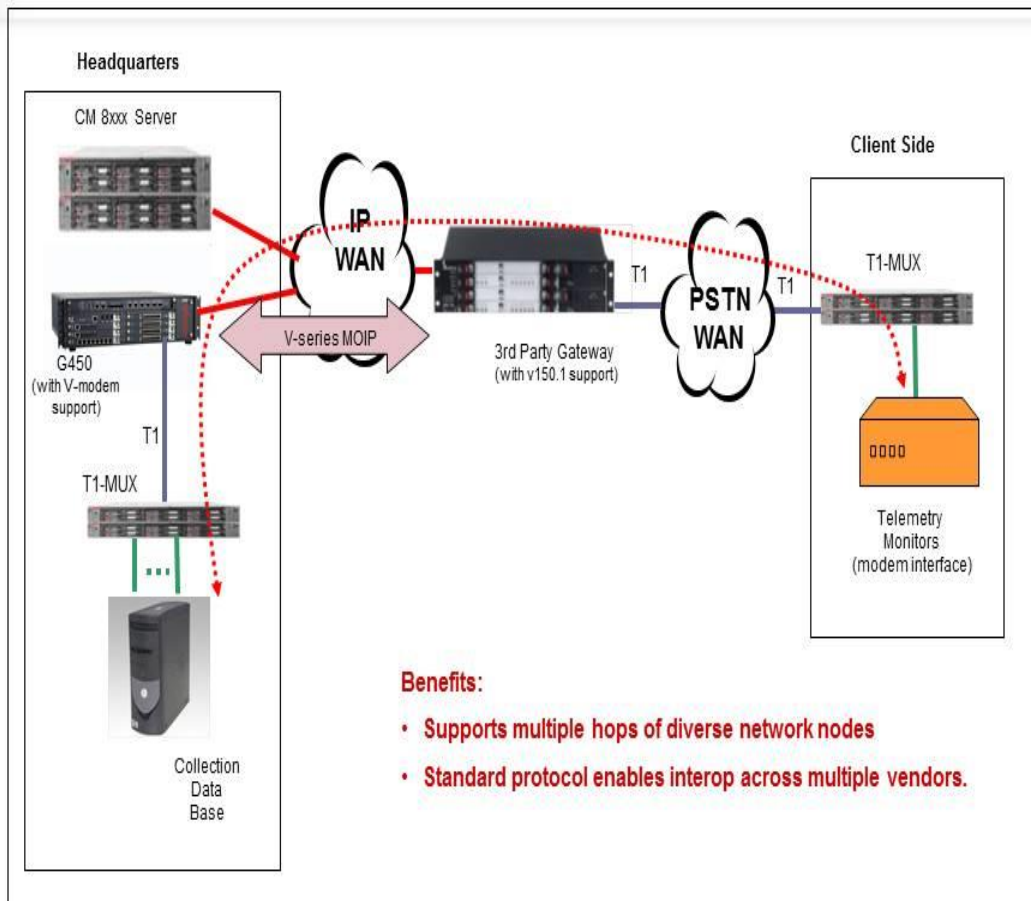
- Resource Distribution
- Energy Monitoring
- Water Management
- Medicine
- Electrical Energy Providers

Customers will now have better opportunities to send telemetry data over their converged communications systems.

Figure 3 illustrates a commercial application where the customer has a data application (e.g., telemetry) where a V.150.1 capable gateway transports PSTN traffic into the CM/SM managed SIP network. With CM Release 6.3 (Release 6.0.1 for government support), the G450 is capable of supporting V.150.1. With CM Release 6.3.2, the G430 is capable of supporting V.150.1. Thus, incoming SIP media streams can be routed directly to the G450/G430 towards the host side data application server. The following application limitations should be noted:

- The modulations supported are V.32, V.34, V.90, and V.92.
- V.42 error correction is not supported.
- The gateways support the digital side of the V.90/V.92 specification.

Avaya Aura™ CM/Gateway Support for Telemetry Monitoring with V150.1 Modem-over-IP



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Figure 3: Avaya Aura CM/Gateway Support for Telemetry Monitoring with V-Series Modem-over-IP

6 Technology Summary

6.1 Gateway Protocol Architecture for Modem-over-IP Application

Figure 4 illustrates the protocols supported by the gateway firmware associated with supporting the DSP algorithms involved with the Modem-over-IP application. The left side of the picture shows that the DSP data pumps will handle the TDM-based modem signals and process the training sequence and demodulate the incoming data. Optionally (refer to the ‘Futures Section in 4.14) there may be processing functions for ‘error correction’ and ‘data compression’. This information is passed up toward the MoIP Application layer.

The gateway firmware will then present the information to the stack on the right side of this diagram for presentation into a Modem-over-IP conversion. A few important aspects of this are summarized:

- For VBD and audio codec presentation, the media stream payload information is differentiated by separate RTP payload types (defined later in this paper).
- The telephony call progress events associated with data and fax (such as answer tones), is sent out-of-band in accordance with the RFC 2833 protocol as a separate RTP payload type.
- The State Signaling Exchange (SSE) protocol allows the application to notify the peer Modem-over-IP entity as to the current operational mode the MoIP application is operating in (such as ‘Audio’, ‘VBD’, ‘Modem Relay’). This information is sent over a separate RTP payload type.
- For the V.150.1 modem relay application, the actual data is conveyed using an SPRT protocol packet. This packet is identified by a separate RTP payload type identifier. It does NOT reside within an RTP packet.

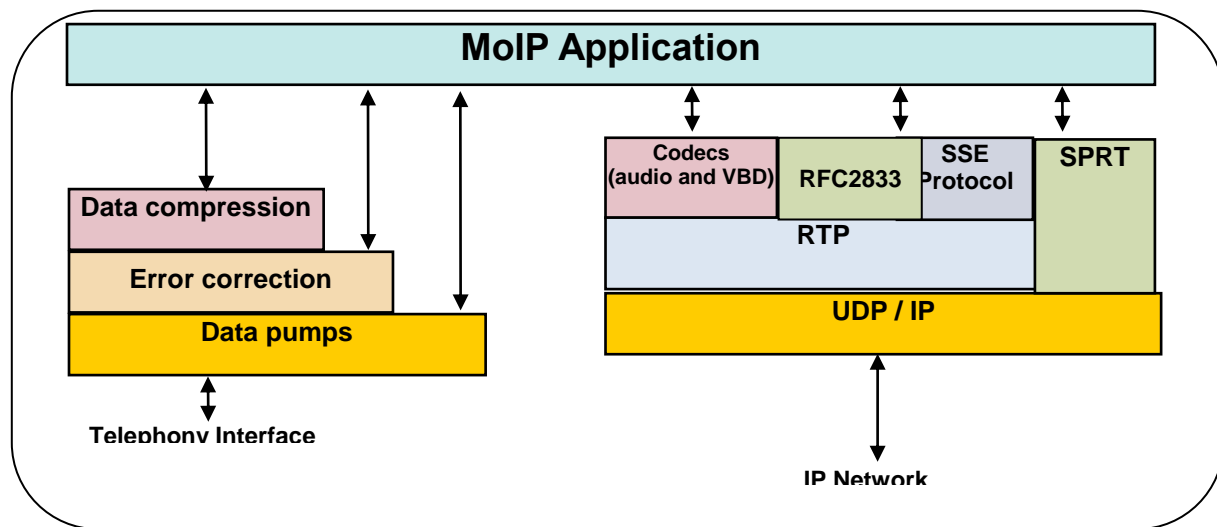


Figure 4: Gateway Protocol Architecture for Modem-over-IP Application

Figure 5 summarizes the four types of SPRT payload packet types. Note that the secure phone communication is transported over TC3 as dictated by SCIP-216.

TC No	Title	Descriptionn
0	Unreliable, unsequenced transport	Used for acknowledgements only
1	Reliable, sequenced transport	Transport channel used for data
2	Expedited, reliable, sequenced transport	Transport channel used for control/signaling messages. Data transported in this channel is expedited to the peer user relative to data that is transported in TC1.
3	Unreliable, unsequenced transport	Transport channel for sequenced data that does not require reliable delivery. (Secure phone data uses this channel)

Figure 5: SPRT Transport Channel Types

6.2 CM to Gateway Media Services Capabilities Exchange

Upon registration of a G450/G430 media gateway with Communication Manager, the CM software will query the gateway to report its new media services capabilities. In this way, CM will discover the capabilities, allowing for interoperation with gateways that have different versions of hardware and firmware.

6.3 CM to Gateway Provisioning & Signaling Flow Sequence

Figure 6 illustrates how two communications systems would handle peer-to-peer signaling to exchange an Offer/Answer and how CM would then send H.248 directives down to its G450/G430 for setting up a V.150.1 capable call connection:

- ▶ Step 1: Upon H.248 registration, the gateway will send Codec feature capabilities exchange (CAPEX).
- ▶ Step 2: CM will offer v150mr as a latent capability within the SDP Offer/Answer during initial call setup, over the SIP trunk to the peer communication device (shown as Nortel in this example).
 - The SDP media attribute description typically has at least 8 lines, which is much more complex than a typical SIP VOIP call.
 - Specify RTP payload type ID for SSE, RFC2833, & SPRT
 - Specify SPRT window size and payload size
 - Specify V.150.1 parameters such as modulation, call disc mode, compression, delay, and version.
- ▶ Step 3: CM will send negotiated capabilities down as a H.248 call request to the gateway
- ▶ Step 4: The transition from voice to modem is managed by the GW using the same UDP port that was used for voice (port sharing).

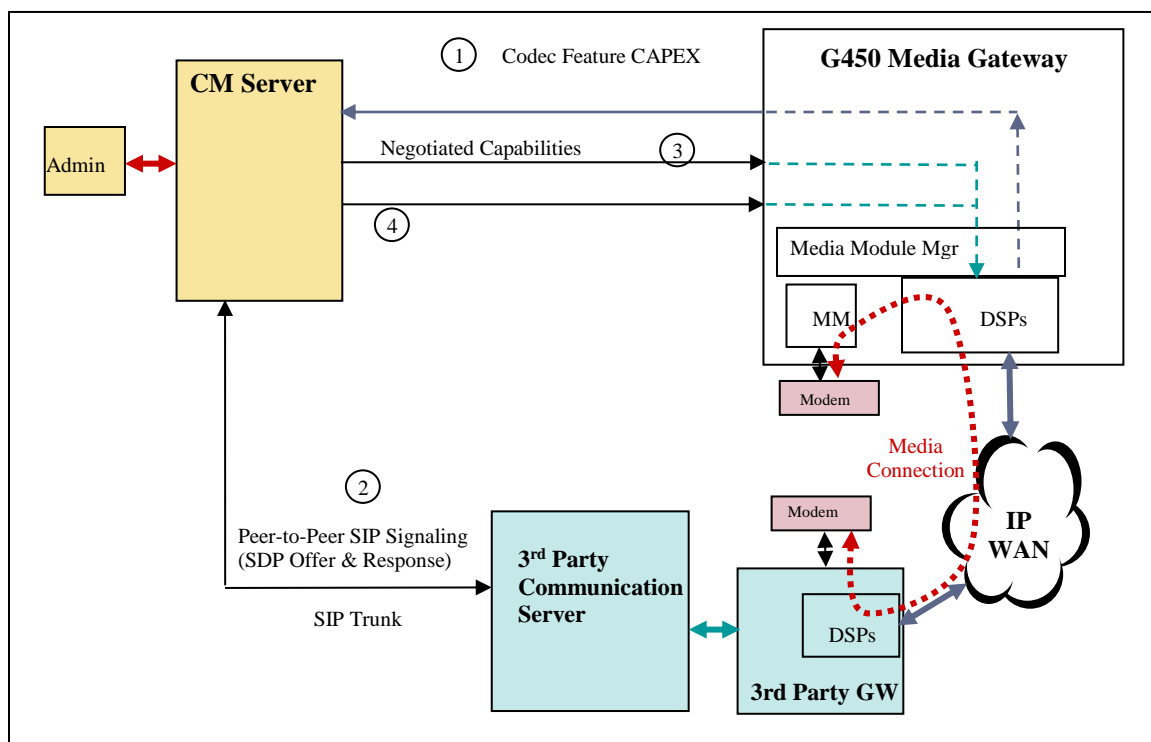


Figure 6: V.150.1 Provisioning & Signaling Flow Sequence

7 DSP Hardware Overview

7.1 DSP Card Support for the G450 Gateway

The G450 Media Gateway supports the ability to mix DSP boards of different types (MP160, MP80, MP20).

Table 3 illustrates the possible combinations of that are acceptable in a G450 chassis configuration. Note that the MP10 is not supported.

The slot placement of the MP cards shall be as follows:

- Gateway Hardware Vintage 1 – MP160 cards may be placed in Slot 2 and Slot 4.
- Gateway Hardware Vintage 2 – MP160 cards may be placed in Slot 1 and Slot 2.

Note: the hardware vintage can be determined using the ‘show platform main’ command.

Combination of Cards	MP80 Card	MP 20 Card	MP 160 Card
Combination # 1	-	-	2
Combination # 2	-	2	1
Combination # 3	2	-	1
Combination # 4	1	1	1

Table 3: G450 Mix & Match MP Card Combinations

Note: The MP160 may be deployed in Slots #2 and #4

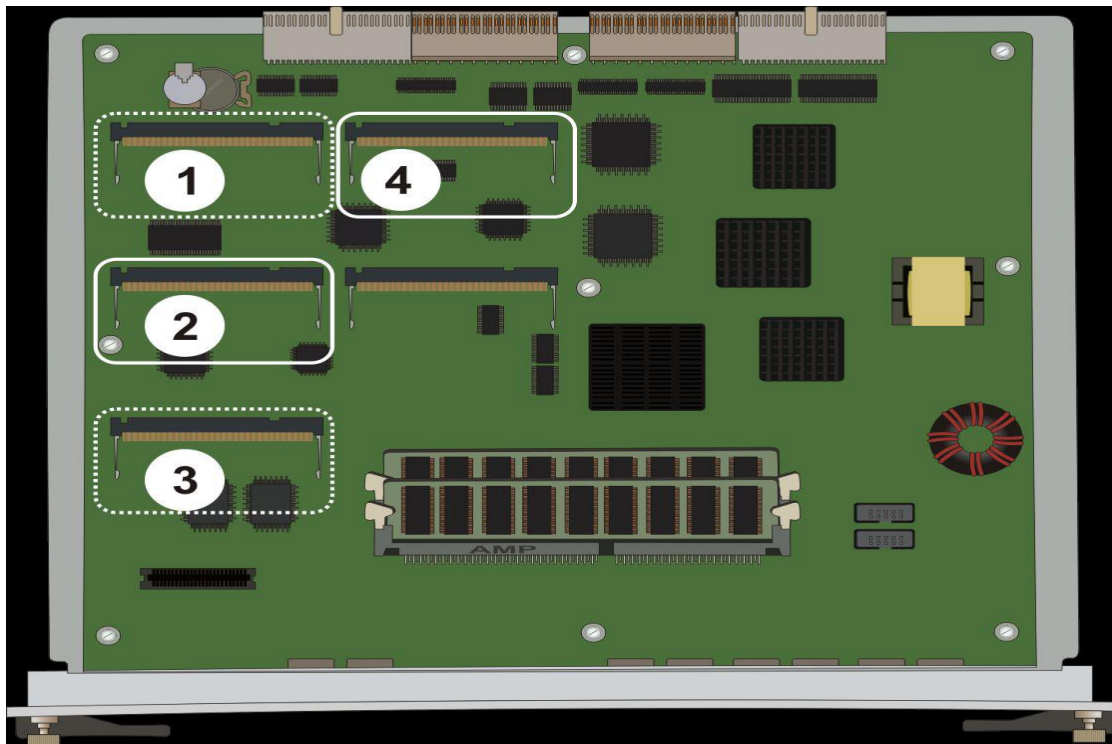


Figure 7: Location of MP160 Slots in G450 Vintage 1 Hardware

Note: The MP160 may be deployed in Slots #1 and #2

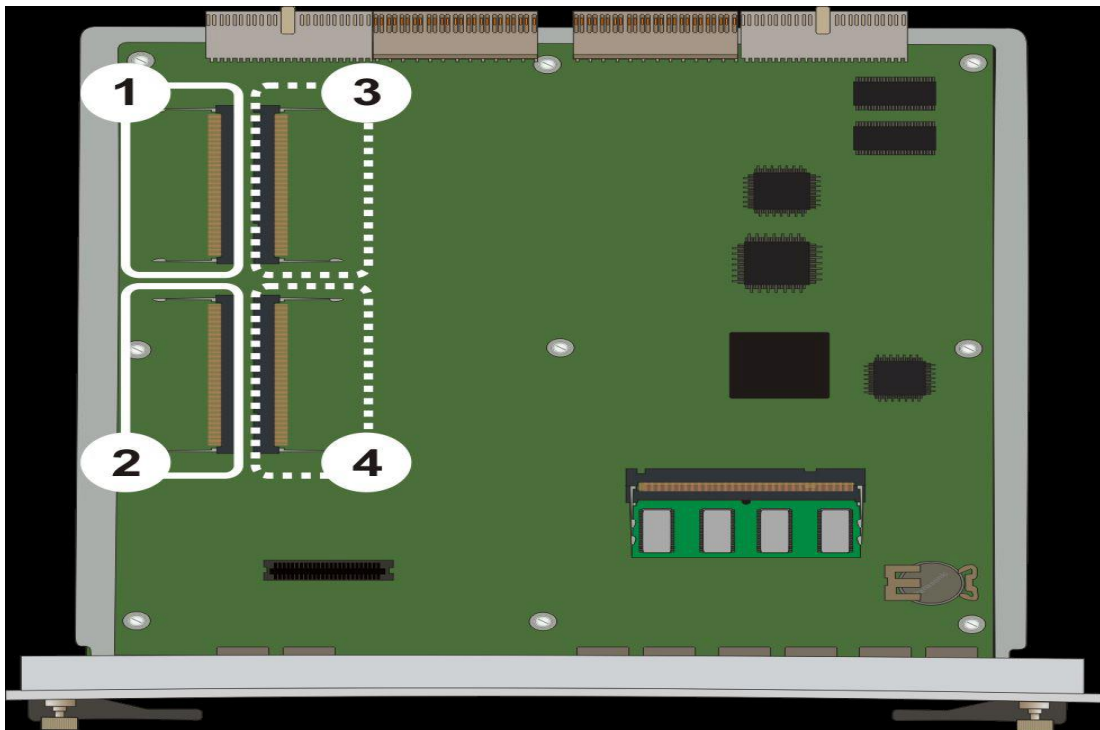


Figure 8: Location of MP160 Slots in G450 Vintage 2 Hardware

7.2 DSP Card Support for the G430 Gateway

The G430 Media Gateway supports the ability to select a DSP board of the following types (MP120, MP80, MP20). The new DSP board, MP120, is capable of supporting 120 channels for either voice or V.150.1 data.

7.3 DSP Point Cost Characteristics

7.3.1 MP 160 DSP Point Cost Characteristics

The MP160 will have a point capacity of 4800 points.

Types of calls are assigned DSP processing “point costs”:

- 30 points for G.711, G.729, G.726, T.38
- 40 points for V.150.1

V.150.1 capable calls are allocated as a 40 point cost at time of the audio phase origination

- Neither CM nor the Gateway can predict whether user of a Government Secure phone (eg, vIPer) would go “secure” or not during a given call session. Therefore, all calls that are negotiated for latent capability for V.150.1, will be allocated at 40 point cost at audio phase origination.

The G450 will allocate the “next” call to the MP card that has the highest percentage of spare capacity

On average (for mix/match of an MP80 + MP160):

- 33% of the calls assigned to MP80
- 67% of the calls assigned to MP160

7.3.2 MP 120 DSP Point Cost Characteristics

The MP120 will have a point capacity of 3600 points.

Types of calls are assigned DSP processing “point costs”:

- 30 points for G.711, G.729, G.726, T.38
- 30 points for V.150.1

Since the point cost is the same for all media applications, there is no need to provide further explanation of point cost engineering for the G430 product.

7.3.3 MP 80 DSP Point Cost Characteristics

The MP80 will have a point capacity of 2400 points.

Types of calls are assigned DSP processing “point costs”:

- 30 points for G.711, G.729, G.726, T.38
- This module DOES NOT support V.150.1

7.4 Media Gateway Channel Allocation for the G450 Gateway

The algorithm considers both of the following issues when determining which DSP core should be used to allocate a DSP channel:

- Load Balancing
- Fragmentation

Load Balancing:

- Channels are allocated so that the percentage of capacity used is evenly distributed across all DSP cores.
- Helps distribute the DSP load required to process calls across multiple cores.
- Reduces the number of call failures in the event of a core crash.

Fragmentation:

- To ensure successful call placement, core allocation must consider "fragmentation."

- For example, fragmentation would occur if all 4 cores had 20 capacity points remaining and a request was made for a V.150 call requiring 40 points. In this case the call would be denied.
- If instead, 2 of the 4 cores had 40 capacity points remaining, and the other 2 were full, the V.150 call would be granted.

7.5 DSP Channel Usage Discussion for the G450 Gateway

Types of calls are assigned DSP Processing “point costs”:

- 30 points for G.711, G.729, G.726
- 40 points for V.150.1

Point capacity per MP card

- MP80 = 2400 points
- MP160 = 4800 points

Assumption is that a typical customer will have either

- One MP80 card and One MP160 card
- Two MP160 cards

Typical customer has less than 10% modem calls

- Government application must be V.150 ready for all connections involving secure terminal devices.
- There are customers (e.g. Energy Monitoring for power utility companies) that require nearly all calls to be modem ready.

The table in **Figure 9** provides a general engineering aid for guiding the traffic design for DSP provisioning in the G450 media gateway.

Homogeneous traffic allows one to declare an absolute max capacity:

- For 2 MP160 cards: can support 320 channels voice (non-V.150.1)
- For 2 MP160 cards: can support 240 channels V.150.1

For the “mixed MP80/MP160 usage:

- Voice (Non-V.150) is allocated at roughly 33% for the MP80 and 67% for the MP160.
- DSPs start to become saturated around 200 voice channels.
- If CM tries to set up a V.150 call but there are no longer enough resources available (i.e. all MP160 channels are in use but there are still MP80 channels available), the gateway will inform CM that there are INSUFFICIENT RESOURCES – NO MOIP_RESOURCES AVAILABLE so that CM knows that there are still resources available non-V.150 calls.

Gateway DSP Deployment				
	2- MP160	MP80 & MP160	2-MP80's & MP160	
All voice (non-V150)	320	240	320	Note: Absolute max with homogeneous traffic
All v150	240	120	120	
Voice + V150	50V, 202 v150	50V, 86 v150	50V, 95 v150	Note: The traffic distribution shows voice-centric applications and illustrates how much V.150 traffic is allowed.
	100V, 165 v150	100V, 53 v150	100V, 70 v150	
	150V, 127 v150	150V, 20 v150	150V, 45 v150	
	200V, 90 v150	200V, 0 v150	200V, 20 v150	
	250V, 52 v150		250V, 0 v150	
	300V, 15 v150			
<div>Getting Saturated</div> <div>Getting Saturated</div>				

Figure 9: DSP Channel Usage Capacity Table

8 End to End Flow Diagrams for V-Series Modulation Exchange

This section has been provided to show a summary of the end-to-end signaling between two modem devices (classical modems or secure phones which utilize modem signaling during the transition to secure mode). This summary is extremely helpful when the user is examining trace captures while doing diagnostic work.

The key for each of these diagrams is as follows:

M1 = "modem 1"

G1 = "Gateway 1"

G2 = "Gateway 2"

M2 = "modem 2"

In these diagrams, Modem M1 is the calling device and Modem M2 is the destination for this call. The answering modem M2 responds with an ANS signal with V.32 or and ANSam signal with the V.34, V.90, and V.92 modulation schemes. These modem "answer" signals are relayed by means of RFC 2833 over the IP fabric between gateways G1 and G2. Once the calling device recognizes the answer signal, the modulation signaling will be sent to Gateway G1, which will initiate a SSE exchange.

Upon a successful SSE handshake, the gateways G1 and G2 will exchange the V.150 control events as follows:

- * SSE Event Sent (Moved to Modem Relay)
- * SSE Event Received (Moved to Modem Relay)
- * SPRT INIT Sent
- * SPRT INIT Received
- * SPRT MR_Event Sent
- * SPRT MR_Event Received
- * SPRT CONNECT Sent
- * SPRT CONNECT Received

Once the SPRT connection has been established between the two gateways, the user data may be exchanged as V.150.1 modem relay packets.

Section 10 shows this exchange as part of a summary of the Avaya Aura Communication Manager and G450/G430 gateway.

8.1 Establishing Modem Relay with V.32 Modem

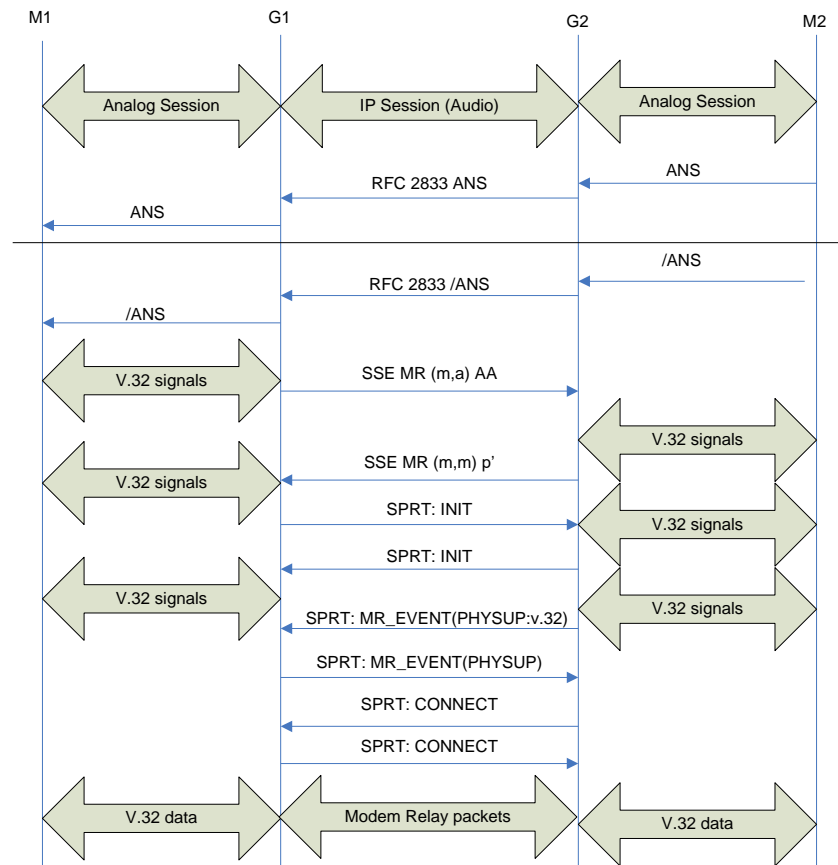


Figure 10: Establishing Modem Relay with ITU V.32 Modem

8.2 Establishing Modem Relay with V.34 Modem

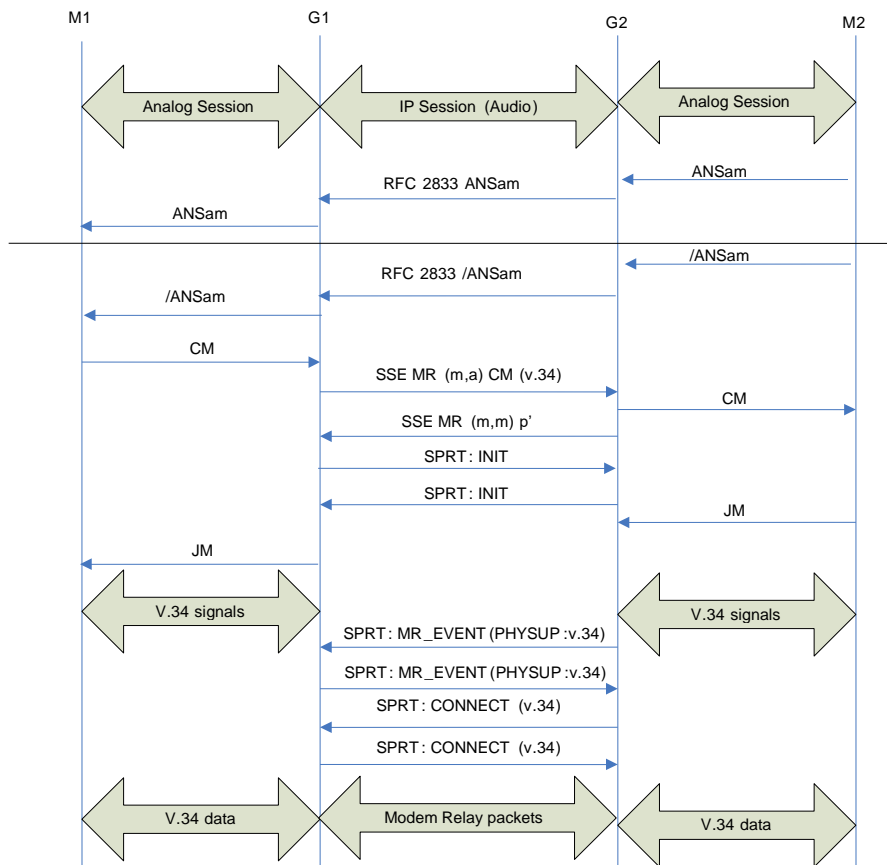


Figure 11: Establishing Modem Relay with ITU V.34 Modem

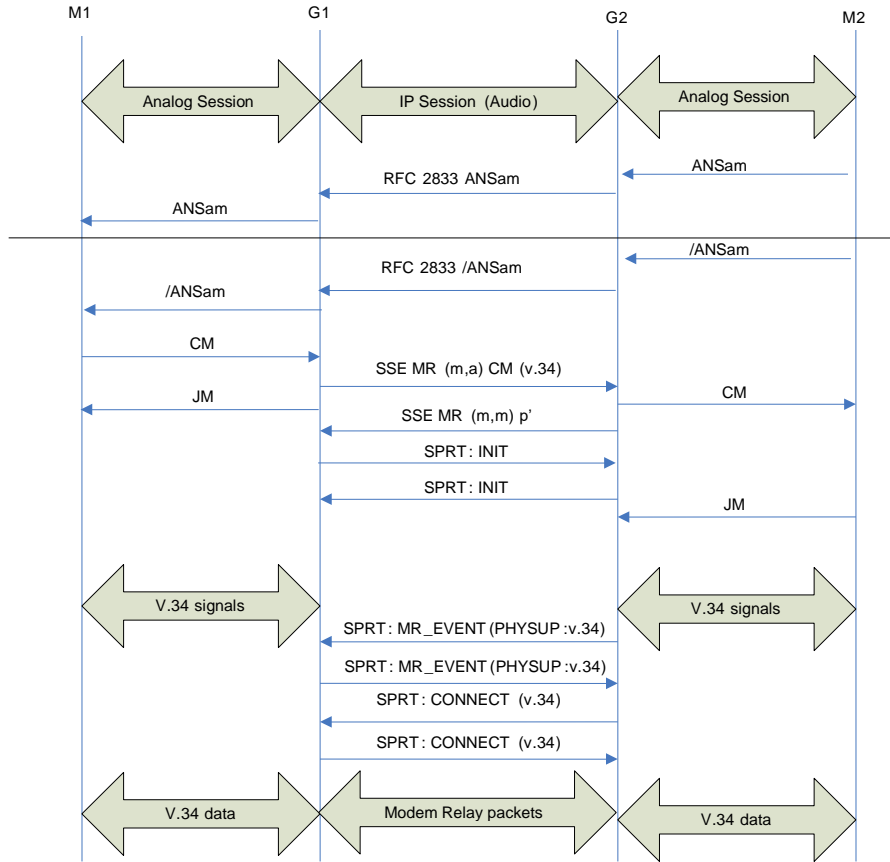


Figure 12: Establishing Modem Relay with ITU V.34 Modem with no JM_INFO Message Sent from G2 Gateway

Note: When the originating modem, M1, sends a CM message indicating that the Modulation modes are a subset of Common Modulation Set "G" (values 1=V.34, and 3=V.32) then the receiving gateway, G2, is not required to send a SPRT:JM_INFO(v.34) message in return. So it is the responsibility of gateway, G1, to immediately return a JM message back toward modem M1.

8.3 Establishing Modem Relay with V.90 Modem

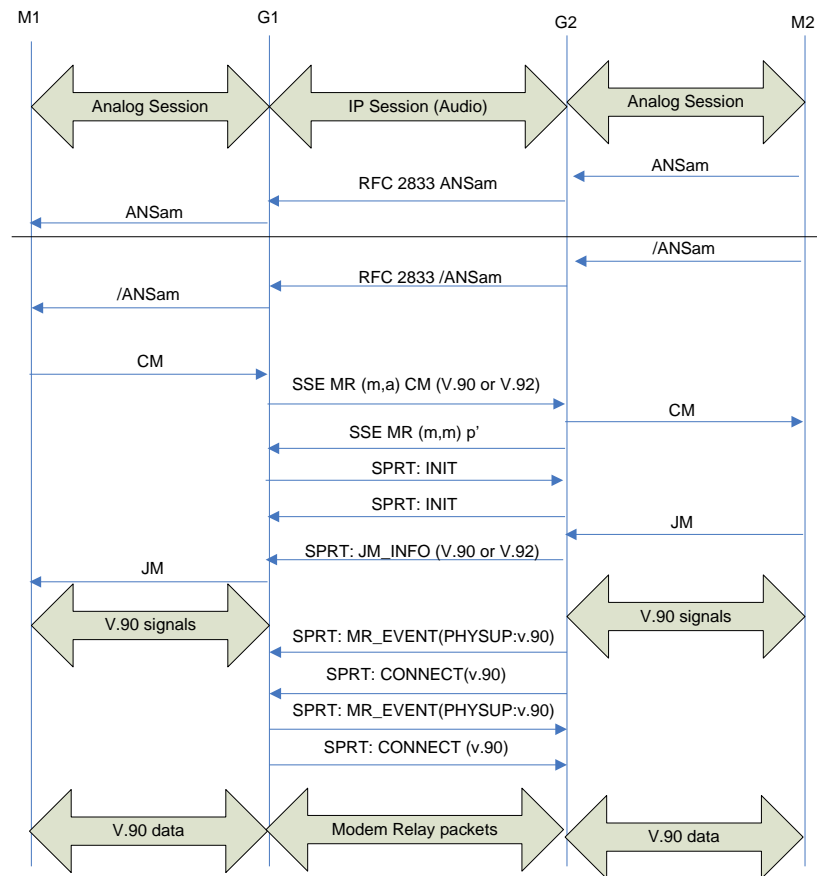


Figure 13: Establishing Modem Relay with ITU V.90 Modem

8.4 Establishing Modem Relay with V.92 Modem

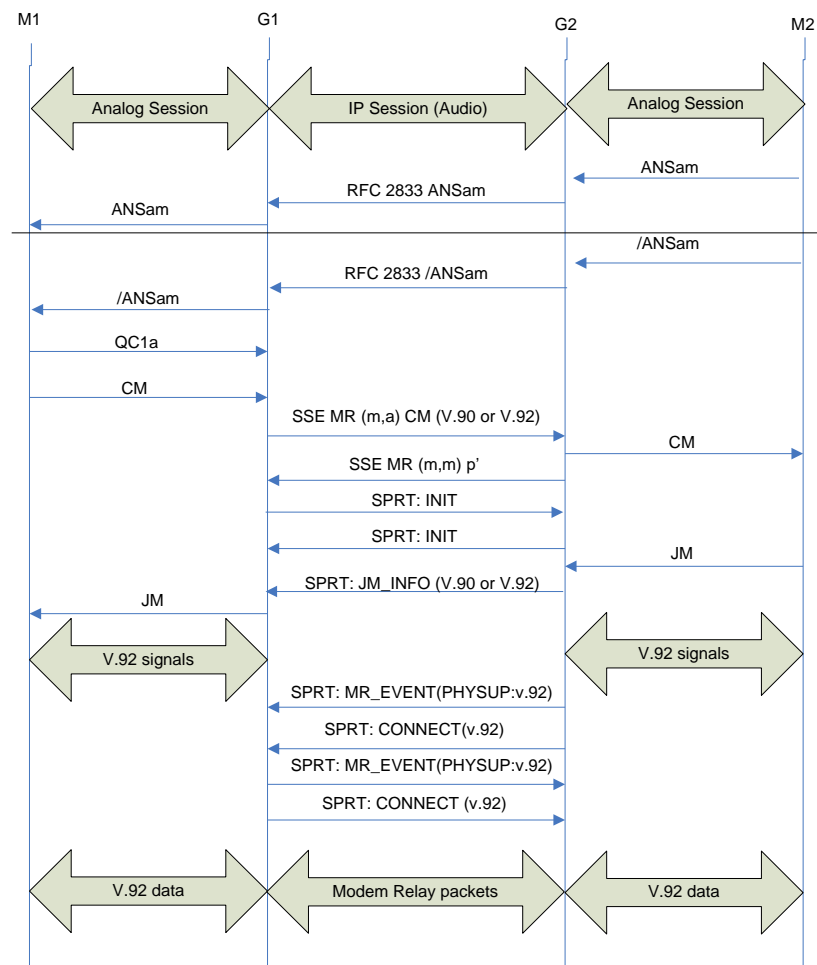


Figure 14: Establishing Modem Relay with ITU V.92 Modem

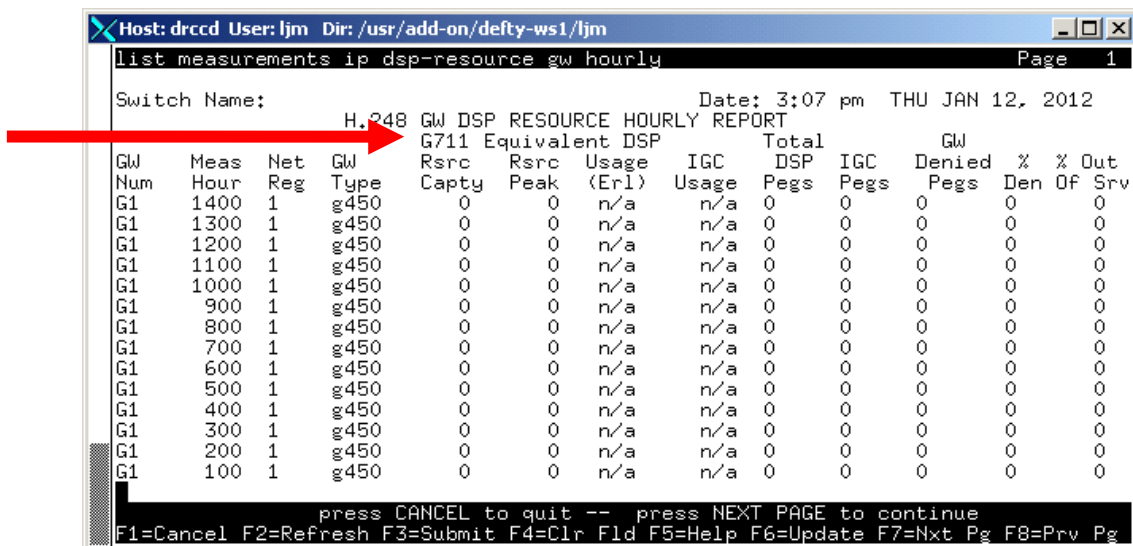
9 List Measurement SAT Performance Tool

CM keeps track of statistics for voice and fax calls over IP that involve the gateway.

Note: Our existing Voice codecs use a DSP point cost of 30 per call. The V.150 protocol will require a point cost of 40 per call.

CM shall accumulate MOIP call statistics along with the present statistics being collected for VOIP/FAX calls.

- The technician may invoke the measurement for gateways with:
 - 'list measurements ip dsp-resource gw<interval>'
 - 'list measurements ip dsp-resource summary'
- The following statistics will be gathered for v150mr calls and aggregate into G711 equivalent usage:
 - DSP Resource Capacity (Max) supported for a given network region
 - Peak DSP resource per measurement period
 - Overall DSP usage per measurement period
 - IGC usage per measurement period
 - Total DSP pegs per measurement period
 - Total IGC pegs per measurement period
 - Gateway call denied per measurement period
 - Percentage calls denied per measurement period
 - Percentage out-of-service events per measurement period
- CM will update the usage by:
 - Using a point cost of 40 for standard v150mr calls
 - Using a point cost of 60 for v150mr calls that use V.42 error correction



Host: drccd User: ljm Dir: /usr/add-on/defty-ws1/ljm

list measurements ip dsp-resource gw hourly Page 1

Switch Name: H.248 Date: 3:07 pm THU JAN 12, 2012

G711 Equivalent DSP

GW Num	Meas Hour	Net Reg	GW Type	Rsrc Capty	Rsrc Peak	Usage (Erl)	IGC Usage	Total DSP Pegs	IGC Pegs	GW Denied Pegs	% Den	% Out Of Srv
G1	1400	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	1300	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	1200	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	1100	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	1000	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	900	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	800	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	700	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	600	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	500	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	400	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	300	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	200	1	g450	0	0	n/a	n/a	0	0	0	0	0
G1	100	1	g450	0	0	n/a	n/a	0	0	0	0	0

press CANCEL to quit -- press NEXT PAGE to continue

F1=Cancel F2=Refresh F3=Submit F4=Clr F1d F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Figure 15: List Measurement Report (Existing Design)

10 List Trace SAT Diagnostic Tool

CM provides some diagnostic trace tools which are part of the SAT interface. When enabled, the list trace tool provides a step-by-step summary of the individual events for a call connection. In CM release 6.2-FP2, there is the inclusion of V.150.1 event reporting as part of the standard trace report list.

For tracing an individual V.150 call, there will be multiple uplink events that report a call.

An example of a call would be as follows:

- * SSE Event Sent (Moved to Modem Relay)
- * SSE Event Received (Moved to Modem Relay)
- * SPRT INIT Sent
- * SPRT INIT Received
- * SPRT MR_Event Sent
- * SPRT MR_Event Received
- * SPRT CONNECT Sent
- * SPRT CONNECT Received
- * SSE Event Received (Cleardown)

The strings for SSE events are defined by the following set:

These are defined in a 'courier new font' to preserve proper display formatting.

Strings:

Comment Notes:

-----	-----
TX SSE:m(CM)	moved to Modem Relay due to V.8 CM detection
TX SSE:m(AA)	moved to Modem Relay due to V.32 AA detection
RX SSE:m(p')	moved to Modem Relay because the peer also did
TX SSE:a(time out)	moved to Audio state due to a time out
RX SSE:a(Cleardown)	moved to Audio due to local modem hangup

The event strings for SPRT (data protocol exchange) are defined by the following set.

Example strings:

TX SPRT:INIT(NECRxCH=0, ECRxCH=1, XID=0, data=0x080) with data type bitmap	tx init
RX SPRT:INIT(NECRxCH=0, ECRxCH=1, XID=0, data=0x080) with data type bitmap	rx init
TX SPRT:MR_EVENT(RRNEG, 1) renegotiation	initiated rate
RX SPRT:MR_EVENT(RETRN, 2) retrain	responded to

```

TX SPRT:MR_EVENT(PHYSUP, V.34, 28800, 28800, 3200, 3200)      V.34 with
tx/rx bit & symbol rates

RX SPRT:MR_EVENT(PHYSUP, V.32, 9600, 9600)                    V.32 with
only tx/rx bit rates

TX SPRT:CONNECT(V.34, 0, 0, 0, 28800, 24000, 0, 0x0001)
V.34, tx/rx 28800/24000, data type

TX SPRT:CONNECT(V.32, 0, 0, 1, 9600, 9600, 0, 0x0001)
V.32, V.42, 9600/9600

RX SPRT:JM_INFO(0x8XXX, 0xeXXX, 0xbXXX)
Call Function, PCM modem avail, PSTN access

TX SPRT:CLRDOWN(Physical Layer Release, 0x00, 0x00)

```

10.1 Verbose Mode for List Trace

The standard ‘list trace’ command has been expanded to offer an option that will provide a “verbose” (expanded) set of event recording.

The command syntax is:

- List trace station ext [gw-diag <n>]
- Where ‘n’ = 0 for default output
- Event reporting is limited to 70 character line width, with a maximum of 4 lines in the particular event that is being reported.
 - Note that most events will be a single line.

11 Types of Secure Phone Devices

11.1 General Dynamics

11.1.1 Sectera® vIPer™ and TalkSECURE™ vIPer™ Universal Secure Phone

These telephones are both IP and PSTN (analog) capable. Sectera telephones provide Type 1 (SCIP-230) secure speech suitable for U.S. and foreign governments.

TalkSECURE models support commercial-grade AES (SCIP-231) encryption but are otherwise similar to Sectera, both physically and in terms of SIP and V.150.1 protocols.

The U.S. government considers the Sectera vIPer to be a Controlled Cryptographic Item (CCI) requiring delivery and accounting through COMSEC channels. Once a key has been loaded and activated with a PIN, the device becomes classified to the level of the key so that any handling must be attended by someone with a U.S. security clearance equal to or greater than that of the key.

Gateway compatibility with the PSTN-capable vIPer is expected at the analog media module interface (MM711, MM714, MM714B, MM716) for both a-law and μ -law. When the telephones are engaged in secure speech across an IP connection, the gateway will also exhibit compatibility at the V.150.1 DCE level, i.e. the gateway's V.150.1 conversion of the telephone's modem signal must result in a secure connection and speech with a far-end.

Figure 16 and Table 4 provide some summary information for the vIPer models.



Figure 16: General Dynamics Sectera vIPer

General Dynamics Part Number	Description
VIPS1000XA	Type 1 US National Standard PSTN vIPer (upgradeable to VoIP SIP Protocol)
VIPS1000XAPTT	Type 1 US National Standard PSTN/Upgradeable to VoIP SIP Protocol w/ PTT Handset
VIPS1000XB	Type 1 UK National Standard PSTN vIPer (upgradeable to VoIP SIP Protocol)
VIPS1000XC	Type 1 Canadian National Standard PSTN vIPer (upgradeable to VoIP SIP Protocol)
VIPS1000XD	Type 1 Australian National Standard PSTN vIPer (upgradeable to VoIP SIP Protocol)
VIPS1000XZ	Type 1 NZ National Standard PSTN vIPer (upgradeable to VoIP SIP Protocol)
VIPS5000XA	TalkSECURE Standard PSTN vIPer (AES Only) (upgradeable to VoIP SIP Protocol)

Table 4: General Dynamics vIPer Models

11.1.2 Sectera® Wireline and TalkSECURE™ Wireline Terminal

The General Dynamics Wireline products provide SCIP adaptation for ordinary analog telephones and commercial fax machines. The adapted devices remain in the analog domain (they're not VoIP-capable). Gateway compatibility with Wireline-adapted devices should be identical to that of the PSTN-capable vIPer.

Figure 17 and Table 5 provide some summary information for the Wireline Terminal models.



Figure 17: General Dynamics Sectera Wireline Terminal

General Dynamics Part Number	Description
1DGT1907XASF	Type 1 US National SWT w/Secure Fax
ISDN1907XASF	Type 1 US National SWT1 (w/ ISDN Modem & Adapter Kit) w/ Secure Fax
1DGT1907XBSF	Type 1 UK National SWT w/Secure Fax
1DGT1907XCSF	Type 1 Canadian National SWT w/Secure Fax
1DGT1907XDSF	Type 1 Australian National SWT w/Secure Fax
1DGT1907XZSF	Type 1 NZ National SWT w/Secure Fax
1DGJ9907XASF	TalkSECURE Wireline (AES Only) w/Secure Fax

Table 5: General Dynamics Wireline Terminal Models

12 Fax Operation in a CM-6.2+ V.150.1 (MoIP) Environment

Since fax machines and modems used a common subset of signals, there is a need for an intermediate call discrimination phase. During this phase, RFC2833 is used to relay signals until a decision can be made as to whether the call involves a fax or modem. If a fax determination is made, the call will transition to fax mode in the same manner as that done in CM 6.1 and earlier. Note that the transition to T.38 implies a signaling step involving CM call control (not shown in the figures below). Transition to Avaya fax relay or Avaya fax pass-through modes are carried out immediately by the media processor.

For the purposes of this section, fax machines generally fall into 2 categories, those that support V.8 (high speed), and those that do not (low speed , i.e. standard G3) . The two categories employ different signals to start the fax session and therefore the handling of the call differs depending on what kind of machine is calling or answering.

The Avaya G450/G430 fax relay implementations only supports low-speed fax relay (high-speed V.34 support may be added in the future). Because of this special handling must be performed to prevent the fax machines from starting a V.8 session (similar to what is described in T.38 Annex F.3).

Similarly, media gateways fall into two categories, those that support V.8 fax relay, and those that only support low-speed T.30 fax relay. Since it is not completely clear how other vendors' gateways in either of these categories will handle fax calls in a standard-based MoIP environment, care must be taken to ensure calls remain in low-speed mode successfully.

The combination of the need to support both V.8 and non-V.8 machines, and the need to force the fax machines to low-speed, and the need to support the interoperation of T.38 with other vendors' gateways all contribute to fax handling methods described in the following sections.

One of the most practical advantages in using V.150.1 is the ability to differentiate between a high speed fax machine and a high speed modem. Without V.150.1, any ip-codec-set that is configured to allow both fax and modems, will end up using the modem pass-through protocol. A V.34 modem in a fax machine and a V.34 modem present the same calling/answer tone. The Avaya Modem Pass-Through protocol is proprietary and will not be recognized by other Vendor's equipment and the call will fail. The V.150.1 modem protocol is an internationally interoperable modem protocol that can distinguish between a V.34 fax tone and a V.34 modem tone. The real value is being able to consistently choose T.38 for fax and V.150.1 for modem calls. Here is an example of a branch with a single gateway that requires both fax and modem support. The far-end SIP gateway is a non-Avaya device that also supports the V.150.1 modem protocol.

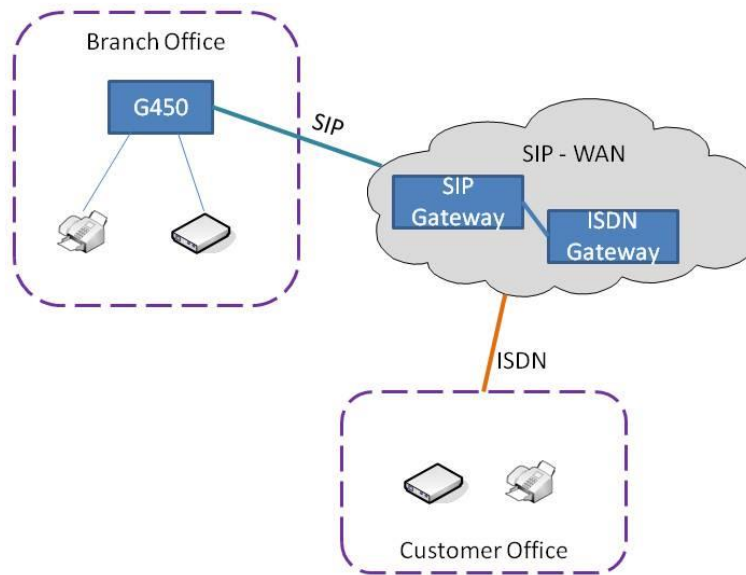
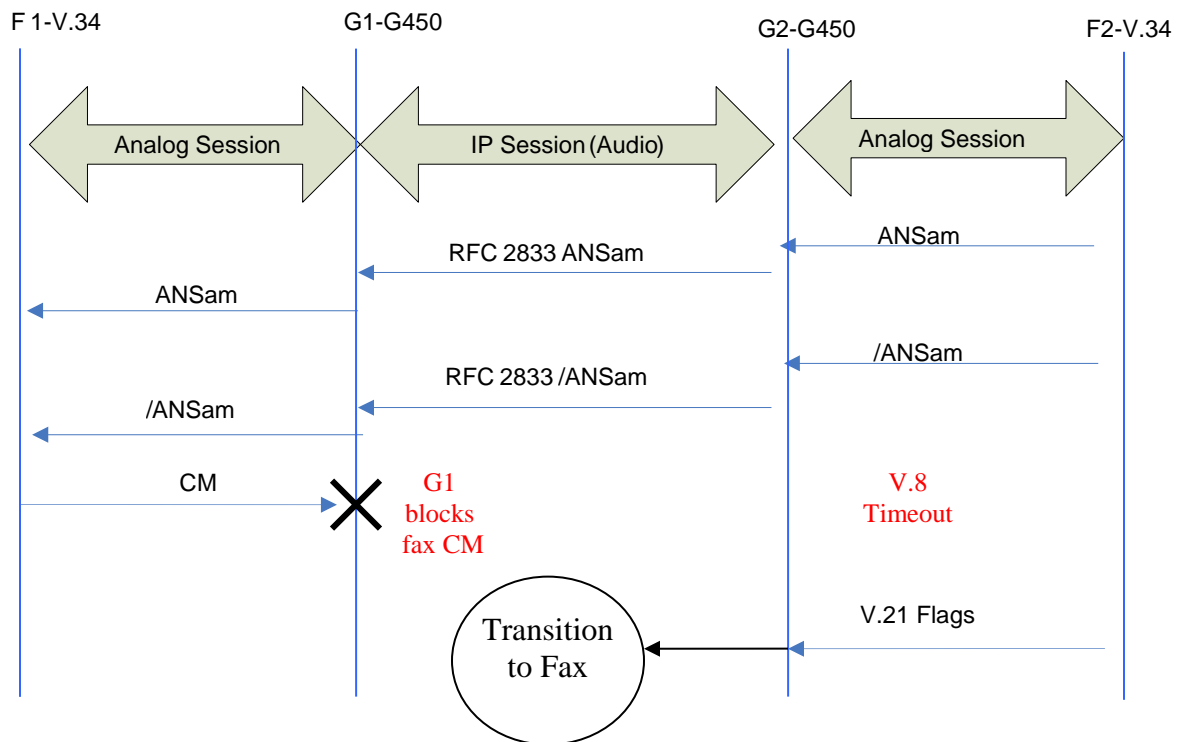


Figure 18: Coexistence of a T.38 fax and V.150.1 modem in a single Network Region

12.1 V.8 Fax Calling, V.8 Fax Answer

The answering fax machine F2 sends ANSam, which is relayed from gateway G2 to gateway G1 (and thus F1) by way of RFC2833. The calling fax machine responds to ANSam with the CM signal. Within the CM signal is contained a function which will indicate fax. Upon detecting the fax function, gateway G1 shall suppress the CM signal, and not relay it to the other gateway (G2). Fax machine F2 will wait for CM until the V.8 timeout period is expended, and then fall back to a low speed T.30 session and send V.21 flags (as part of the DIS sequence). Gateway G2 shall detect V.21 flags and begin the fax session. This is illustrated in **Figure 19**.



Note: A V.34 capable Fax machine responds to an incoming V.34 answer tone with a CM (Call Menu) message.

Figure 19: V.8 Fax Calling and V.8 Fax Answering

In the case when gateway G1 is another vendors' gateway which supports V.8 fax relay, G1 may decide to send an SSE packet to indicate that it has entered Modem Relay mode and that the CM is the stimulus. Therefore the Avaya gateway G2 must inspect the function field within the SSE(CM) packet and if fax is indicated, must drop/ignore this packet and not regenerate the CM signal. This is illustrated below in **Figure 20**.

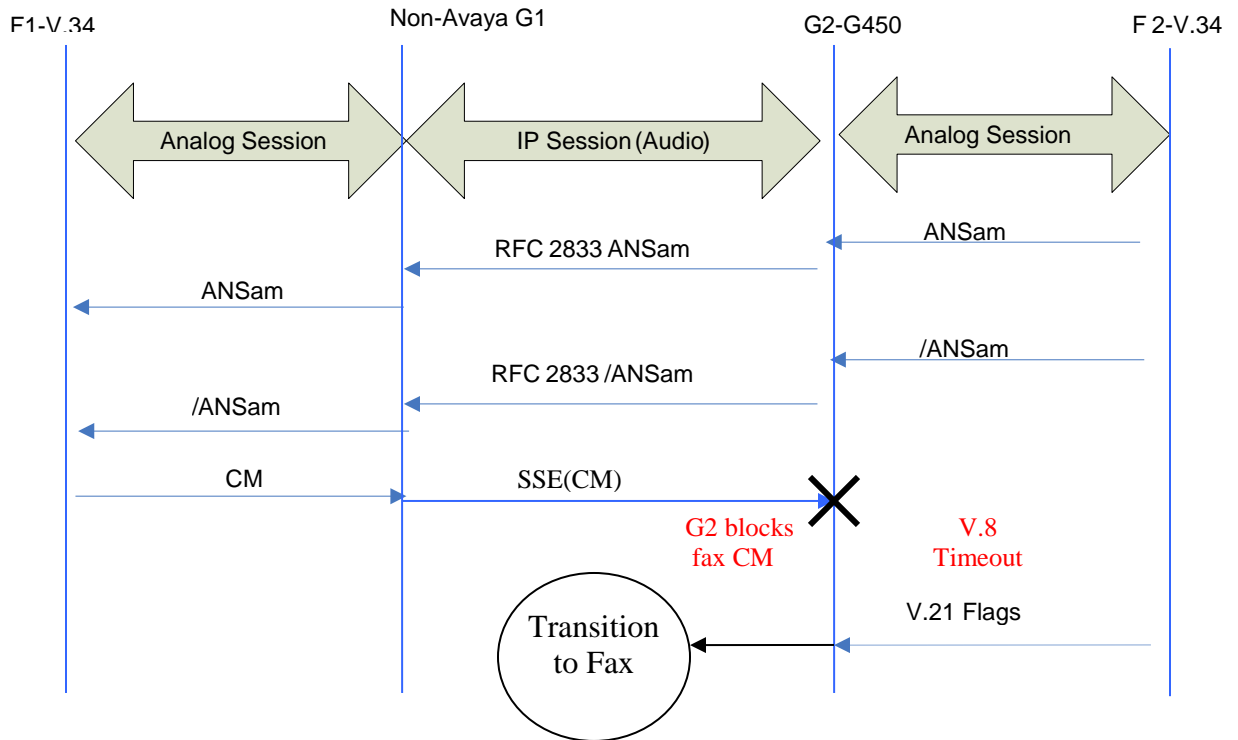


Figure 20 – Non-Avaya Gateway on V.8 Calling-side, Avaya on V.8 Answer-side

12.2 Low-Speed Fax Calling , V.8 Answer

In this scenario, since the calling fax machine does not support V.8, it will not send a CM signal. The answering V.8 machine will fall back to low speed fax upon timeout waiting for CM, and then proceed to send V.21 flags. Gateway G2 will detect V.21 flags and start the fax session. This is illustrated in **Figure 21**.

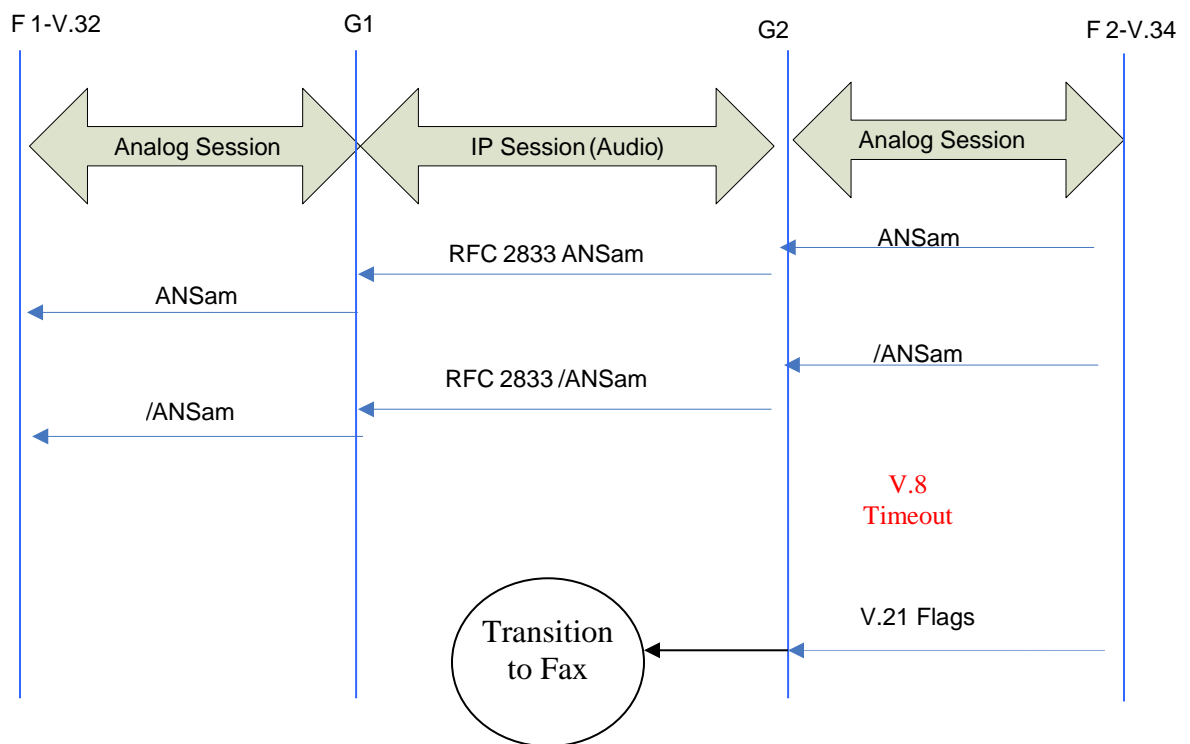


Figure 21 - Low-Speed Fax Calling, V.8 Answer

12.3 Low Speed Fax Answer

In this case, the fax type of the calling side is irrelevant. The answering fax machine sends CED. Gateway G2 is unable to determine whether the connected machine is a fax machine or the V.25 answer tone from a modem. Therefore, G2 relays the 2100Hz notification via RFC2833. Since ANSam is not sent to the calling side, no V.8 session is attempted even if the calling side is V.8 capable. The answering fax eventually sends V.21 flags, which triggers the start of fax mode.

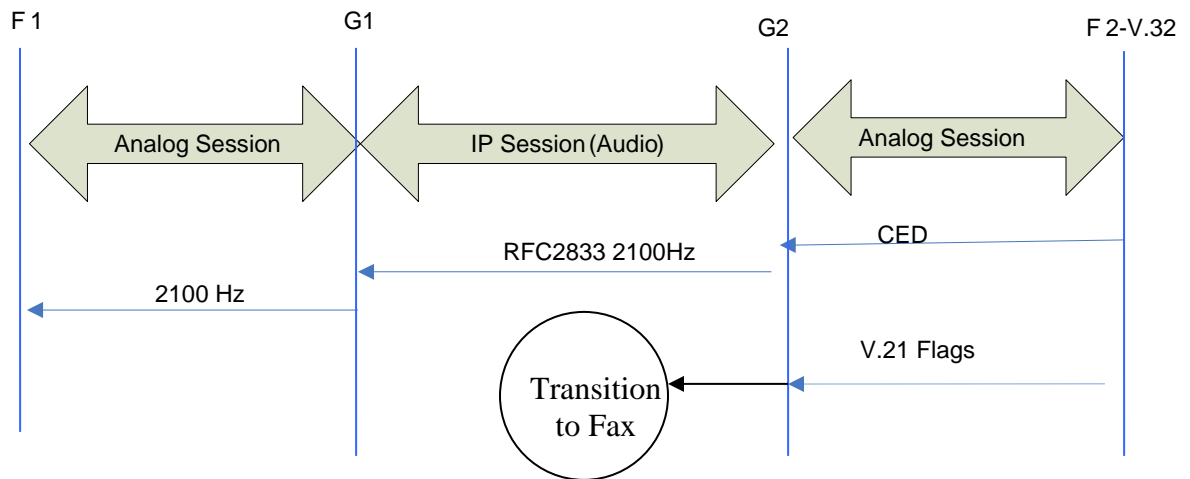


Figure 22 – Low Speed Fax Answer

13 Types of V.150 Compatible Gateways

13.1 AudioCodes

13.1.1 Mediant 3K

This is a high density SIP Trunk Gateway which can terminate up to 84 T1 trunks and convert the streams to SIP trunks. The product supports up to three DS3 mini-SMB coax interfaces or a single optical OC-3 interface. The TP-6310 trunk gateway module, and all components of the chassis may be duplicated for high availability. This product became GA in February, 2011 as part of the High Density SIP Trunk Gateway family that Avaya offers as a SIP solution integrator.



Figure 23: Mediant 3000 High Availability (duplicated TP-6310 modules)

13.2 NET Gateway

The NET VX1800 is a medium size SIP gateway which has a resident SIP registrar and communication controller for branch offices that is UCR compliant.

A summary of features is as follows:

- 3U chassis that contains six slots
 - 8 port analog module
 - T1/E1 digital modules in sizes (1 port, 2 port, 4 ports)
- SIP registrar
- Communications Controller with dial plan routing
- Router & Firewall
- Supports G.711, G.723, G.729, T.38 fax



Figure 24: NET VX1800: Intelligent Voice Gateway

14 Types of Commercial Data Modems

The following models of commercial data modems have been tested in our development labs.

- V.92 MultiTech model MT5634ZBA
- V.92 US Robotics model 5686
- V.90 US Robotics model 5686-03
- V.34 US Robotics models 0648 and 0459
- V.32 Paradyne Comsphere model 3820

15 Appendix A: Administration of Communication Manager

15.1 Communication Manager Administration Settings for 'ip codec set'

Page 1 of the 'ip codec set' form supports bearer encryption using AES or SRTP. SSE signaling is encrypted to setup the SPRT session but the SPRT packets are not encrypted at all. This may present security issues because the packets that contain the data are not encrypted. Using Media Encryption is therefore encouraged; either AES or any of the SRTP forms available on the media encryption section on page 1.

135.9.76.39 - PuTTY

change ip-codec-set 6

Page 1 of 3

IP Codec Set

Codec Set: 6

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1:	G.729	y	3	30
2:	G.711MU	n	2	20
3:		-	-	
4:		-	-	
5:		-	-	
6:		-	-	
7:		-	-	

Media Encryption

1: 1 -srtp-aescm128-hmac80

2:

3:

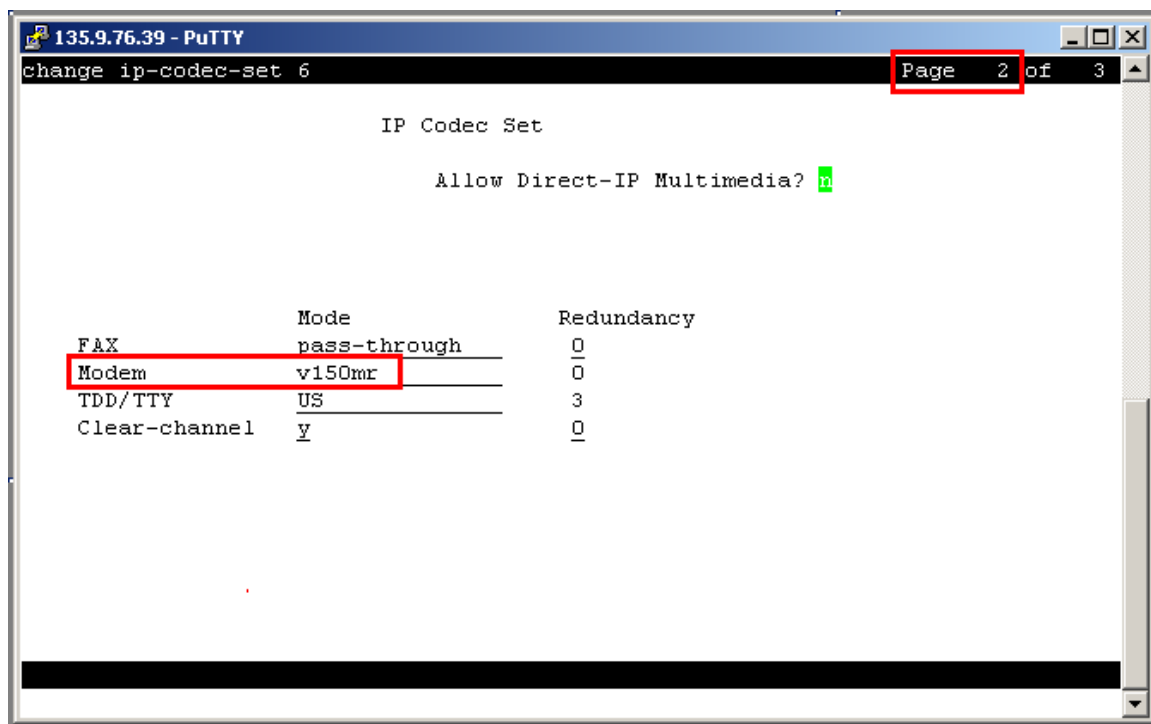
Figure 25: Page 1 of 'ip codec set' SAT Form

The page 2 of the ‘ip codec set’ form was modified to support the ITU standard “Modem over IP” transport protocol.

- New “modem” type: v150mr
 - ‘redundancy’ is display only and is set to ‘0’.

When “v150mr” is selected, a 3rd page of the “ip codec set” is displayed:

- Most customers should be able to use the defaults on Page 3.
- The reasons for modifying “page 3 parameters” are for achieving superior interoperability with certain 3rd party vendor’s equipment.



	Mode	Redundancy
FAX	pass-through	0
Modem	v150mr	0
TDD/TTY	US	3
Clear-channel	y	0

Figure 26: Page 2 of ‘ip codec set’ SAT Form

stinger1.dr.avaya.com - PuTTY

change ip-codec-set 6 Page 3 of 3

V.150 MODEM RELAY PARAMETERS

Note: Maximum SPRT Payload sizes may vary across different gateways dependent on firmware/hardware version

	TC0	TC1	TC2	TC3
SPRT Payload Sizes (bytes)	140	132	132	140
SPRT Window Sizes (packets)	-	32	8	-
SPRT Timer TA01 (msec)	-	90	90	-
SPRT Timer TA02 (msec)	-	130	500	-
SPRT Timer TR03 (msec)	-	500	500	-

SPRT Retransmissions: 3

SSE Repetition Redundancy (packets): 3 V.42 Error Correction? n
 SSE Inter-Packet Interval (msec): 20 V.42bis Data Compression? n
 Modem Relay Preferred (NoAudio)? n

MODULATION MODE

V.32? y V.34? y V.90? n V.92? n

Note: This page will be displayed only if 'v150mr' is set as the "modem type".

Figure 27: Page 3 of 'ip codec set' SAT Form (with v150mr)

A summary of the 'V150mr' parameters is as follows:

Screen Note: "Note: Maximum SPRT Payload sizes may vary across different gateways dependent on firmware/hardware version.

- SPRT TCO Payload Size
 - Range is 140 to 256 bytes
 - Default size is 140 bytes
- SPRT TC1 Payload Size
 - Range is 132 to 256 bytes
 - Default size is 132 bytes
- SPRT TC1 Window Size
 - Range is 32 to 96 packets
 - Default size is 32
- SPRT TC2 Payload Size
 - Range is 132 to 256 bytes
 - Default size is 132 bytes
- SPRT TC2 Window Size
 - Range is 8 to 32 packets
 - Default is 8 packets
- SPRT TC3 Payload Size
 - Range is 140 to 256 bytes
 - Default size is 140 bytes
- SPRT Timer TC1-TA01
 - Ack timer for channel 1

- Range is 50 mSec to 150 mSec (increments of 10 mSec)
 - Default is 90 mSec
- SPRT Timer TC1-TA02
 - Ack Update timer for channel 1
 - Range is 50 mSec to 1000 mSec (increments of 10 mSec)
 - Default is 130 mSec
- SPRT Timer TC1-TR03
 - Retransmit timer for channel 1
 - Range is 50 mSec to 1000 mSec (increments of 10 mSec)
 - Default is 500 mSec
- SPRT Timer TC2-TA01
 - Ack timer for channel 2
 - Range is 50 mSec to 150 mSec (increments of 10 mSec)
 - Default is 90 mSec
- SPRT Timer TC2-TA02
 - Ack Update timer for channel 2
 - Range is 50 mSec to 1000 mSec (increments of 10 mSec)
 - Default is 500 mSec
- SPRT Timer TC2-TR03
 - Retransmit timer for channel 2
 - Range is 50 mSec to 1000 mSec (increments of 10 mSec)
 - Default is 500 mSec
- SPRT-Retransmissions
 - Maximum number of retransmissions for data transfer on channel TC1 and TC2
 - Range is 1 to 32 retries
 - Default is 3 retries
- SSE Repetition
 - Redundant Packets
 - Range is 0 to 3 packets
 - Default is 3 packets
 - Inter-Packet interval
 - Range is 10 to 40 mSec, in increments of 10 mSec
 - Default is 20 mSec
- NoAudio
 - Values are “yes” , “no”
 - Default is “no”
- V42 Error Correction
 - Display only and set to “no”

Notes on administration of the Modulation Mode lines.

- Each field is Boolean with values of either “yes” or “no”.
- The defaults shall be as follows:
 - V.32 is “yes”
 - V.34 is “yes”
 - V.90 is “no”
 - V.92 is “no”
- If the administrator checks “yes” for V.90, CM SAT Administration will automatically fill “yes” for V.34 (this is considered “display only” in this situation.
- If the administrator checks “yes for V.92, CM SAT Administration will automatically fill “yes” for V.34 (this is considered “display only” in this situation, and “yes” for V.90.

Notes on administration of the Modulation Mode lines (Refer to Figure 28:).

- The user may select these modulation modes exclusively or in combination.
- The selection of V.90 or V.92 will always cause the selection of V.34 as well.



USER CASE	V.32	V.34	V.90	V.92
1				
2				
3				
4				
5				

Figure 28: Modulation Mode Selection Options

15.2 RTP Payload Type Assignment

Note: The CM software will select the dynamic RTP Payload Type assignment beginning with value “101”. As an example, if CM were to issue a SIP SDP request for a secure call, it would set the SDP media parameters as follows:

- RFC2833 will be assigned to RTP Payload Type = 101
- V.150 SSE will be assigned to RTP Payload Type = 102
- V.150 SPRT will be assigned to RTP Payload Type = 103

When working with third party V.150 gateways, which offer administrative permissions for these protocols, please set these gateways to “enable dynamic RTP” if configurable.

16 Appendix B: Administration of General Dynamics Sectera vIPer Phones

16.1 General Dynamics vIPer Administration Settings

These secure telephones are both IP and PSTN capable. Sectera telephones provide Type 1 (SCIP-230) secure speech suitable for US and foreign government communications. The remainder of this section provides a summary of the administration required for the SIP mode of operation.

The vIPer administrative name for ‘silence suppression’ is “VAD”. The vIPer default for VAD is “enabled”. By contrast, Avaya CM has silence suppression “disabled” as its default. The vIPer phone and CM should be administered in a consistent fashion.

For the administration guide, please refer to the General Dynamics Sectera and TalkSECURE vIPer Phone Administrator’s Manual.

Each vIPer phone has an embedded web server that uses the HyperText Transfer Protocol (HTTP). You can access the administrative interface from a web browser as long as you know the IP address of the phone. You may directly connect to the LAN interface by connecting a laptop PC via a cross-over Ethernet cable to the phone.

NOTE: Be aware that when you log into the phone, you are suspending its abilities to make or receive calls until you terminate the administrative session!

The General Dynamics vIPer opens up the application by displaying the “System Information” page. There is a navigation pane in the upper left hand corner that offers the following choices:

- System Information
- Phone Settings
- Network Settings
- SCCP Configuration
- V.150 Parameters
- General Settings

16.1.1 System Information

The System Information page provides a current status of the administered setting for the “Phone”, “Network”, “V.150 Parameters”. These are “display-only” and may not be used to change any administered values.

Additionally, the firmware version is displayed.

16.1.2 Phone Settings

The following phone parameters may be administered:

- IP address Display <enabled/disabled>
 - If enabled, the phone will display its IP address.
- Number of Allowed Call Appearances
 - Presently, the phone may only handle one call. So this parameter has no effect.

16.1.3 Network Settings

The following IP address parameters may be administered:

- DHCP <enabled/disabled>
- IP Address
 - Only used if DHCP is disabled
- Subnet Mask
 - Only used if DHCP is disabled
- Default Gateway
 - Only used if DHCP is disabled
- Hostname
 - Set to 'SM domain'

The following DNS Settings may be administered:

- DNS Configuration
 - Set to 'manual'
- DNS Server IP
 - Set to 'DNS IP address'
- DNS Alternate
 - Optional
- DNS Domain Name
 - Set to 'SM domain'

The following VLAN Settings may be administered:

- Voice VLAN Port
 - Set to 'disabled'
- PC VLAN Port
 - Set to 'disabled'

16.1.4 SCCP Configuration

This settings group applies only to the proprietary Skinny Call Control Protocol (SCCP) parameters. These do not apply for an Avaya Aura solution. The following options should be administered:

- TFTP Server IP Address
 - Set to 'disabled'
- Precedence Softkeys
 - Set to 'disabled'

16.1.5 SIP General Settings

The following SIP Configuration parameters should be administered per Figure 29.

GENERAL DYNAMICS
Sectera® vPer™ Phone
Logout

Configuration Options:
System Information
Phone Settings
Ethernet Settings
Network Settings
SCCP Settings
SIP General Settings
SIP Feature Settings
SIP MLPP Settings
SIP Dial Settings
TLS/SRTP Settings
TLS Certificates
V.150 Settings
Voice Settings
Custom Ringtones
Code Upgrade
Change Password
Restore Factory Defaults

SIP General Settings

Proxy Configuration

	Current Setting	New Setting
SIP Registration Proxy	IPv4: 192.168.1.35 IPv6:	192.168.1.35
SIP Outbound Proxy	IPv4: IPv6:	
Disable		
SIP UDP Port	5060	5060
SIP TLS Port	5061	5061
ANAT in Require Header	Disabled	Disabled

SIP Session

	Current Setting	New Setting
SIP Session Refresh Method	UPDATE	UPDATE
SIP Session Expiration {90 -- 10000}	1800	1800
SIP Minimum Session Expiration {90 -- 10000}	90	90

SIP Message Headers

	Current Setting	New Setting
SIP Header User Identification	Directory Number	Directory Number
SIP User-Agent Field	Disabled	Disabled

Date and Time Settings

	Current Setting	New Setting
SNTP Server	IPv4: 216.229.4.69 IPv6:	216.229.4.69
GMT Offset	0	0
Daylight Savings	Enabled	Enabled

QoS Settings

	Current Setting	New Setting
SIP Signaling DSCP	Auto	Auto
SIP Signaling DSCP Value {0 -- 63}	40	40
Default Voice DSCP	Auto	Auto
Default Voice DSCP Value {0 -- 63}	46	46
Web Admin DSCP Value {0 -- 63}	18	18

Submit
Reset Form to Defaults

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Figure 29: General Dynamics SIP General Settings Page

- SIP Registration Proxy
 - Set to ‘IP address for SM SIP-Proxy’
- SIP Outbound Proxy
 - Set to ‘disabled’
- SIP UDP Port
 - Set to ‘5060’
- SIP TLS Port
 - Set to ‘5061’

The following SIP Sessions parameters should be administered:

- SIP Session Timer
 - Set to 'enabled'
- SIP Session Expiration
 - Set to '3600'
- SIP Minimum Session
 - Set to '90'

The following Date & Time parameters should be administered:

- SNTP Server
 - Set to 'IP address for NTP server'
- GMT Offset
 - Set to "Offset to Greenwich Mean Time"
 - For example, set to '-7' for Denver, Colorado
- Daylight Savings
 - Set to 'enabled'

The following DSCP (DiffServe Code Point) parameters should be administered to match the enterprise LAN routing rules. The following are some suggested values:

- SIP Signaling DSCP
 - Set to '40'
- Default Voice DSCP
 - Set to '46'
- Web Configuration
 - Set to '18'

16.1.6 SIP Feature Settings

The following SIP Feature Settings parameters should be administered per Figure 30.

GENERAL DYNAMICS
Sectéra® vPer™ Phone Logout

Configuration Options:

- System Information
- Phone Settings
- Ethernet Settings
- Network Settings
- SCCP Settings
- SIP General Settings
- SIP Feature Settings**
- SIP MLPP Settings
- SIP Dial Settings
- TLS/SRTP Settings
- TLS Certificates
- V.150 Settings
- Voice Settings
- Custom Ringtones
- Code Upgrade
- Change Password
- Restore Factory Defaults

SIP Feature Settings

Line 1 Settings

	Current Setting	New Setting
Directory Number	4020	<input type="text" value="4020"/>
Fully Qualified Display Name	4020	<input type="text" value="4020"/>
Number of Appearances	4	<input type="text" value="4"/>
SIP Authentication User ID	4020	<input type="text" value="4020"/>
SIP Authentication Password	*****	<input type="password" value="*****"/>
Voicemail ID		<input type="text"/>
Line 1 SIP URI: 4020@192.168.1.35		

Line 2 Settings

	Current Setting	New Setting
Enabled/Disabled	Disabled	<input type="text" value="Disabled"/>
Directory Number		<input type="text"/>
Fully Qualified Display Name		<input type="text"/>
Number of Appearances	4	<input type="text" value="4"/>
SIP Authentication User ID		<input type="text"/>
SIP Authentication Password		<input type="password"/>
Voicemail ID		<input type="text"/>
Line 2 SIP URI:		

Early Media Dialing

	Current Setting	New Setting
Early Media Dialing	Disabled	<input type="text" value="Disabled"/>

Call Forwarding Settings

	Current Setting	New Setting
Call Forwarding Agent	Phone	<input type="text" value="Phone"/>
No Answer Ring Count	5	<input type="text" value="5"/>
Reminder Ring	Disabled	<input type="text" value="Disabled"/>

Voicemail Settings

	Current Setting	New Setting
SIP Voicemail Service	Disabled	<input type="text" value="Disabled"/>
Contact Voicemail Server Via:	Registration Proxy	<input type="text" value="Registration Proxy"/>
SIP Voicemail Server	IPv4: IPv6:	IPv4: <input type="text"/> IPv6: <input type="text"/>
Voicemail Key Dial		<input type="text"/>

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Figure 30: General Dynamics SIP Feature Settings Page

The following Line 1 (appearance) parameters should be administered

- Phone Number
 - Set ‘SIP station number’
- Number of Appearances
 - Set to ‘3’
- SIP Authentication
 - Set to ‘enabled’
- SIP User ID
 - Set to ‘SIP station number’
- SIP Password
 - Set for ‘SIP station password’
- Voicemail ID

- Set to 'SIP mailbox number'

The following Line 2 (appearance parameters should be administered following the guidelines presented in Line 1'

The following Call Forwarding Settings should be administered:

- Call Forwarding Agent
 - Set to 'phone'
- No Answer Ring Count
 - Set to '5'

The following Voicemail Settings should be administered:

- SIP Voicemail Service
 - Set to 'enabled'
- Contact Voicemail
 - Set to 'registration proxy for ASM'
- SIP Voicemail Server
 - Set to 'IP address of ASM' or 'IP address of CM for direct SIP trunk'
- Voice Key Dial
 - Set to 'vIPer button to auto-dial message server'

16.1.7 SIP Dial Settings

Note: The default Dial Settings must be changed to accommodate the local dial plan on CM/SM. If you do not administer proper values, the dial plan search will take an extra long time!

The Primary Dialing String should be administered:

- Peer to Peer Dialing
 - Set to 'enabled'
- Interdigit Timeout (in seconds)
 - Set to '4'
- Dial Plan
 - This is user defined, based on the CM dial plan.
 - For example [2-8]xxxx | 9[0-5] | [2-8]xxxx |99xxxxxxxxxxxx

The Secondary Dialing Settings should be administered

- Message Interval
 - Set to '50 ms'
- Digit Duration
 - Set to '100 ms'
- Hook Flash Duration
 - Set to '100 ms'

16.1.8 SIP Security Settings

The following TLS for SIP Signaling channel should be administered:

- TLS Encryption
 - Set to 'disallow' unless a certificate pair is installed.
- Session Lifetime
 - Set to '3600'
- Persist TLS Sessions
 - Set to 'disabled'
- Keep-Alive Interval
 - Set to '120'
- Proceed with Expiration
 - Set to 'enabled'

The following SRTP for the media stream should be administered:

- SRTP Encryption
 - Set to 'disallowed'
- SRTCP Encryption
 - Set to 'enabled'
- SRTP Authentication
 - Set to '32'
- Key Derivation
 - Set to '0'

16.1.9 SIP Codec Settings

The following should be set (if you plan to allow usage of these codecs):

- g.711mu Priority=1 Packet Duration=20ms Jitter Buffer=20ms
VAD=Enabled
- g.726AB Priority=2 Packet Duration=20ms Jitter Buffer=40ms
VAD=Enabled
- g.723.1 Priority=3 Packet Duration=30ms Jitter Buffer=60ms
VAD=Enabled

16.1.10 V.150 Parameters

The following SIP V.150 Common Settings parameters should be administered per Figure 31.

GENERAL DYNAMICS
Sectera® vIPer™ Phone

Configuration Options:

- System Information
- Phone Settings
- Ethernet Settings
- Network Settings
- SCCP Settings
- SIP General Settings
- SIP Feature Settings
- SIP MLPP Settings
- SIP Dial Settings
- TLS/SRTP Settings
- TLS Certificates
- V.150 Settings
- Voice Settings
- Custom Ringtones
- Code Upgrade
- Change Password
- Restore Factory Defaults

V.150 Settings

Common Settings

	Current Setting	New Setting
SSE Reliability Repetition Count:	3	3
SSE Reliability Repetition Interval: {0 -- 5000} ms	20 ms	20 ms
SSE Recovery Repetition Count:	5	5
SSE Recovery Repetition Interval: {500 -- 5000} ms	1000 ms	1000 ms
SSE Recovery Transient Media Interval: {500 -- 9999} ms	2500 ms	2500 ms
TC2 TA01 Timer: {0 -- 9999} ms	90 ms	90 ms
TC2 TA02 Timer: {0 -- 9999} ms	500 ms	500 ms
TC2 TR03 Timer: {0 -- 9999} ms	500 ms	500 ms
Secure Session Bandwidth Limit	41000 bps	41000 bps

SIP Payload Types

	Current Setting	New Setting
Dynamic Payloads:	Enabled	Enabled
Telephone Event Payload: {96 -- 127}	101	101
SSE Payload: {96 -- 127}	97	97
SPRT Payload: {96 -- 127}	98	98
No Audio Payload: {96 -- 127}	126	126
Secure Session SDP Advertisement:	Clear-Setup	Clear-Setup

Submit Reset Form to Defaults

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Figure 31: General Dynamics V.150 Settings Page

We recommend the following settings (use the defaults, where possible):

- SSE Reliability Repetition Count
 - Set to '3'
- SSE Reliability Interval
 - Set to '20'
- SSE Recovery Count
 - Set to '5'
- SSE Recovery Interval
 - Set to '1000'
- SSE Recovery Media
 - Set to '2500'
- TC2 TA01 Timer
 - Set to '90'
- TC2 TA02 Timer
 - Set to '500'
- TC2 TR03 Timer

- Set to '500'
- Secure Session Bandwidth Limit
 - Set to 41000bps

For the SIP Payload Types screen, the following should be administered:

- Dynamic Payloads
 - **Set to 'enabled', so that the phone will follow the SIP Offer from CM when it returns the SIP "200 OK" answer.**
 - Note: When "Dynamic Payloads" is set to 'disabled', the phone will use the provisioned RTP Payload Types shown below AND will NOT adjust these in the classic SDP Offer/Answer protocol exchange.
- Telephone Events Payload
 - Set to '96'
- SSE Payload
 - Set to '97'
- SPRT Payload
 - Set to '98'
- No Audio Payload
 - Set to '126'
- Secure Session SDP
 - Set to 'clear setup'

16.1.11 Code Upgrade

Use for upgrading vIPer firmware.

16.1.12 Change Password

Use to change the vIPer password.

16.1.13 Restore Factory Defaults

Use to reset the factory defaults.

17 Appendix C: Administration of General Dynamics Sectera Wireline Terminal

17.1 General Dynamics Wireline Terminal Administration Settings

This product is an adaptor that is inserted between a conventional analog telephone and the gateway's analog line interface card. The adapted devices remain in the analog domain (they're not VOIP-capable).

Note: For administrative details, please refer to the chapter on General Dynamics vIPer secure phone.

18 Appendix D: Administration of NET VX1800 Gateway

18.1 NET VX1800 Gateway Administration Settings

The NET VX1800 gateway has the ability to receive PSTN trunks (analog and T1), terminate the protocol and convert to SIP trunks which may be forwarded into CM/SM servers. Secondly, the Gateway has a resident SIP registrar and a communication controller (for call routing) that is UCR compliant.

For administration, the first thing to do is to administer a 'trunk group'. As part of this, you will need to declare the codecs being used. Select the **Telephony > Trunk Groups** navigation.

Please refer to Figure 32 as a guideline:

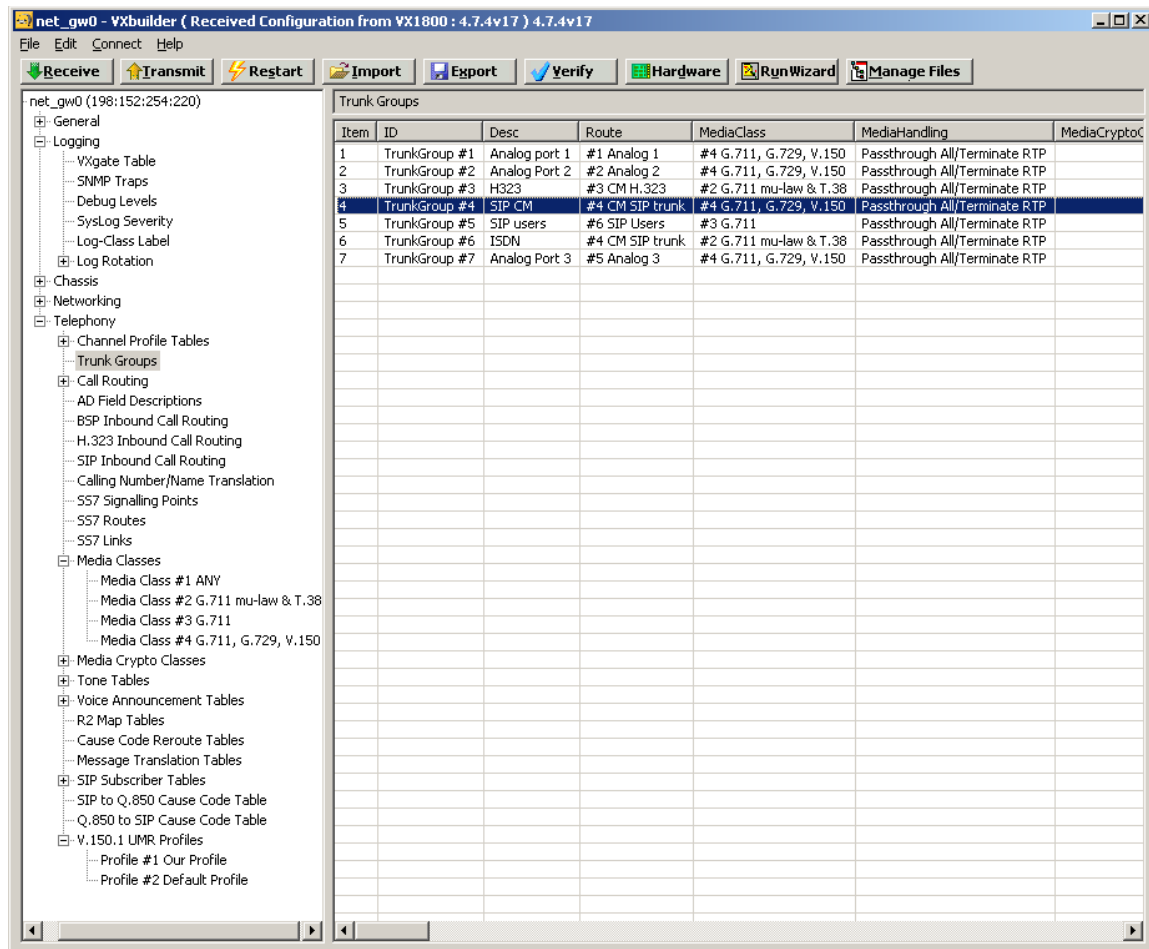


Figure 32: VX1800 Trunk Group Screen

Next select the **Telephony > Media Class** navigation to show how the ‘UMR’ entry is configured for modem relay. See **Figure 33**.

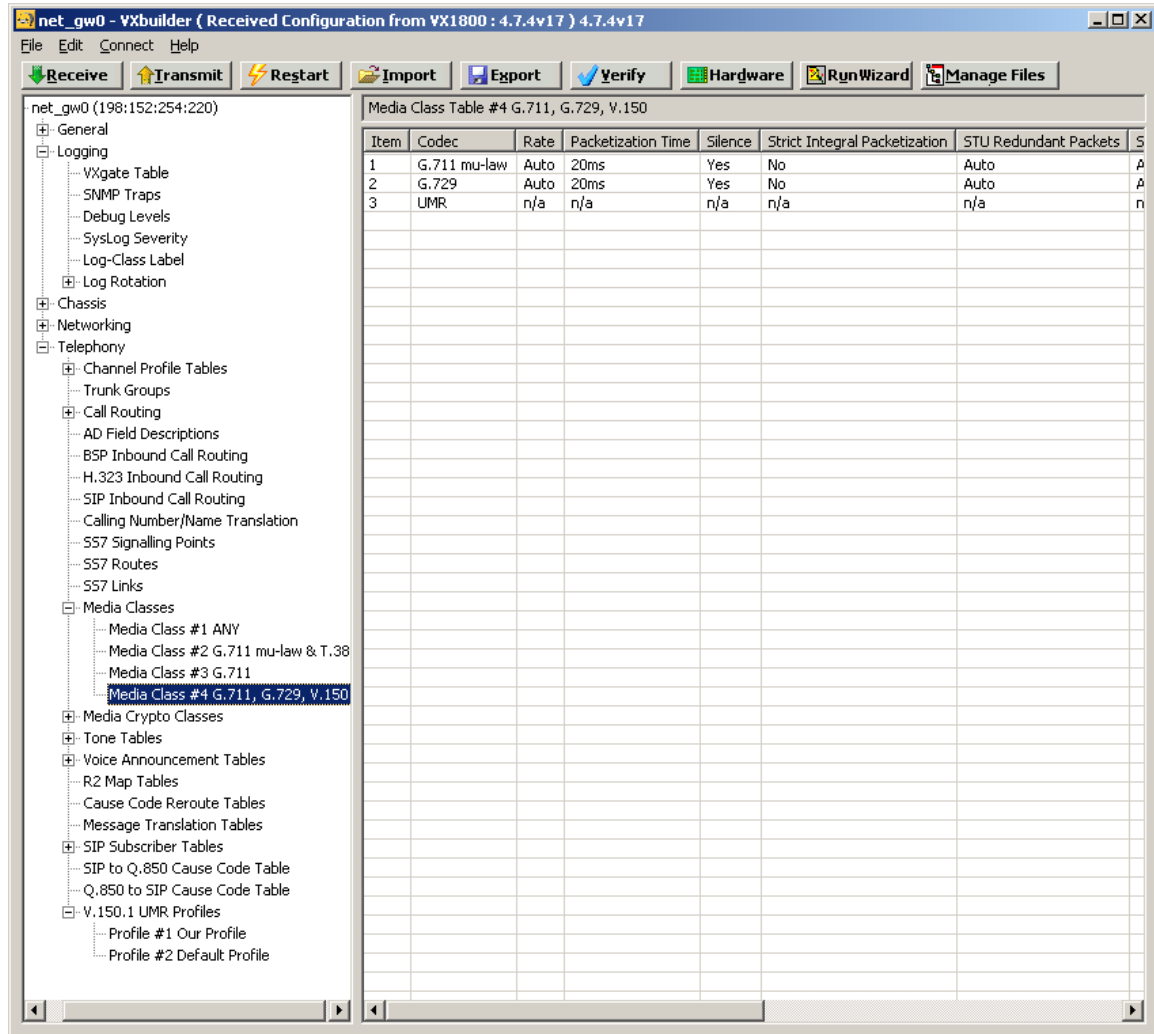


Figure 33: VX1800 Media Class Screen

Next select the **Telephony > V.150.1 UMR Profiles > V.150.1 UMR Profile #n** navigation flow. It is recommended that the default parameters be used for the values of the SPRT timers. Make sure that you assign these values as consistent with page 3 of CM's SAT screen in the 'ip codec set'.

See **Figure 34**.

We recommend setting the following fields:

- TDM: V.8 Enable = 'yes'
- TDM: V.34 Enable = 'yes'
- TDM: V.32 Enable = 'yes'
- SSE: Repetition Count = '3'
- Leave TC1/TC2/TC3 Timers set to the defaults per the CM 'ip codec set' form.

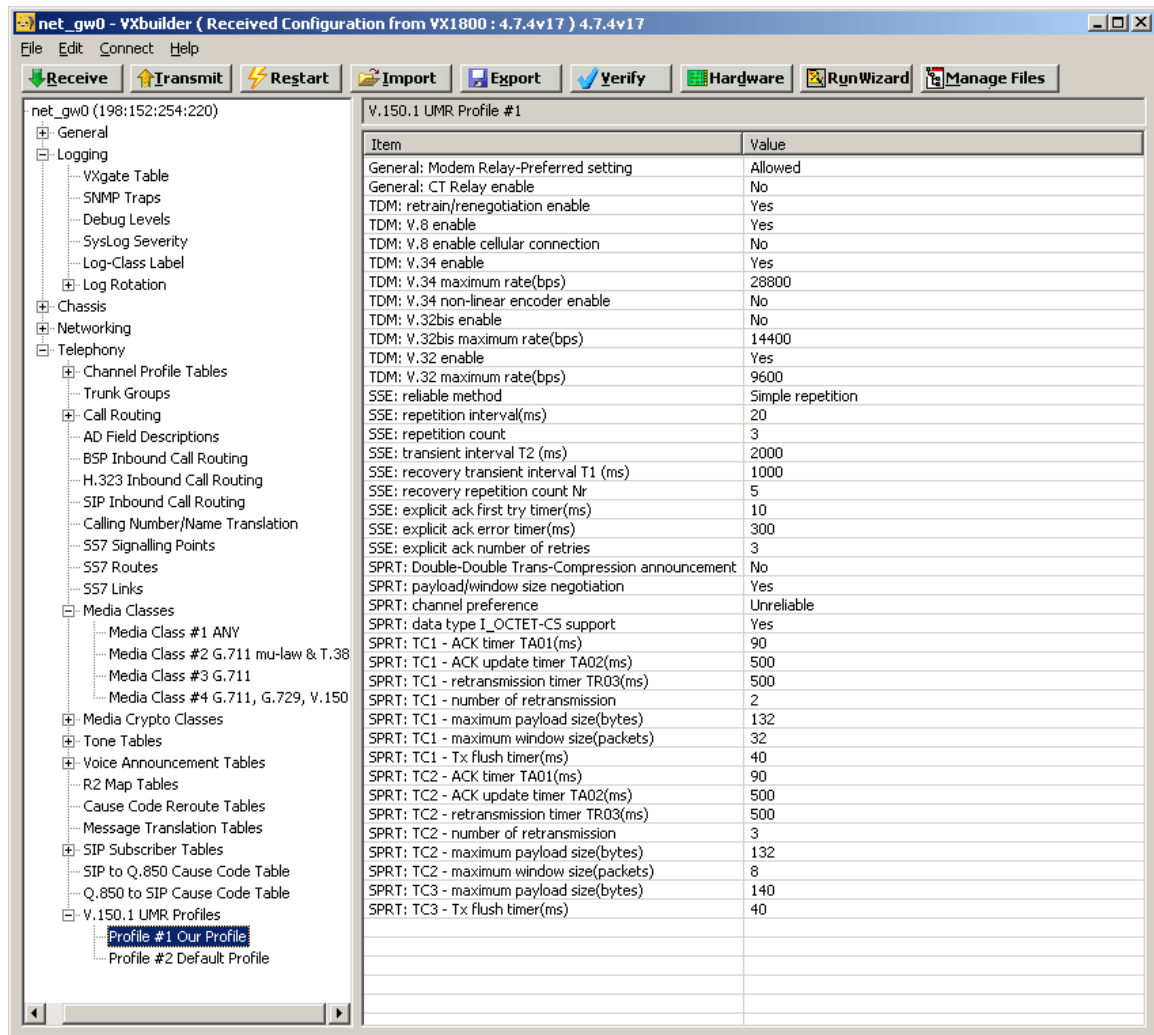


Figure 34: VX1800 V.150 UMR Profile Screen

19 Appendix E: Administration of AudioCodes Mediant 3000 Gateway

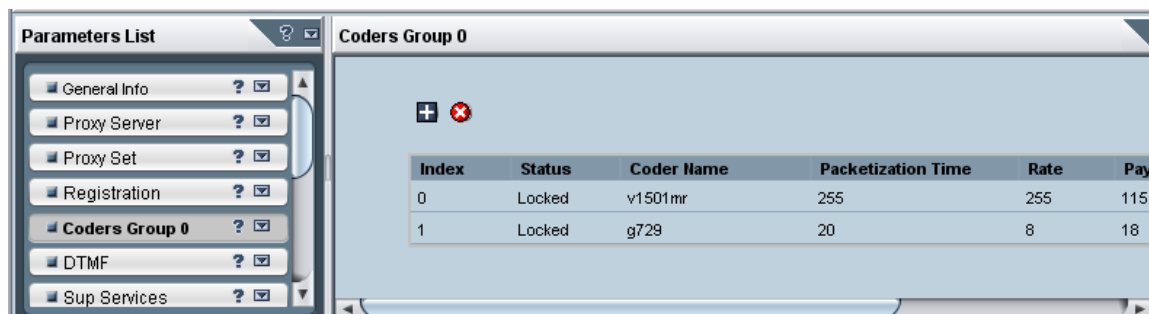
19.1 AudioCodes Mediant 3000 Gateway Administration Settings

The AudioCodes Mediant 3000 gateway has the ability to receive PSTN trunks, terminate the protocol and convert to SIP trunks which may be forwarded into CM/SM servers.

The M3000 gateway product serves as a SIP-based High Density Trunk Gateway. It is typically deployed in a data center/headquarters site to provide cost effective distribution of bundles of PSTN trunks for redirection into a CM/SM managed SIP-based communication network. This M3000 gateway has an interface card (TP-6310) which supports one, two, or three DS3 interfaces. The larger bandwidth interfaces contain bundles of up to 28 trunk circuits. These T1 circuits may support T1-Robbed Bit or ISDN Primary Rate signaling. The M3000 gateway provides the termination of the incoming/outgoing PSTN calls and converts these calls to IP-based calls which support the SIP signaling protocol. The M3000 gateway routes these calls via SIP trunk groups over to Communications Manager or Session Manager.

The V.150.1 Modem Relay feature set was developed for Avaya Government Solutions and is available only with the proper license key.

In the M3K's navigation pane, select **VOIP > SIP>Configuration>SIP Protocol Definitions>Coders Group 0** tab. Add 'v150mr' to the list and administer the RTP payload type to be used with this codec. See the figure below:



Index	Status	Coder Name	Packetization Time	Rate	Pay
0	Locked	v1501mr	255	255	115
1	Locked	g729	20	8	18

Figure 35: M3K Coders Screen

Open the 'Modem Settings' screen (**VOIP > Media Frame>Modem Settings** tab) per the figure below:

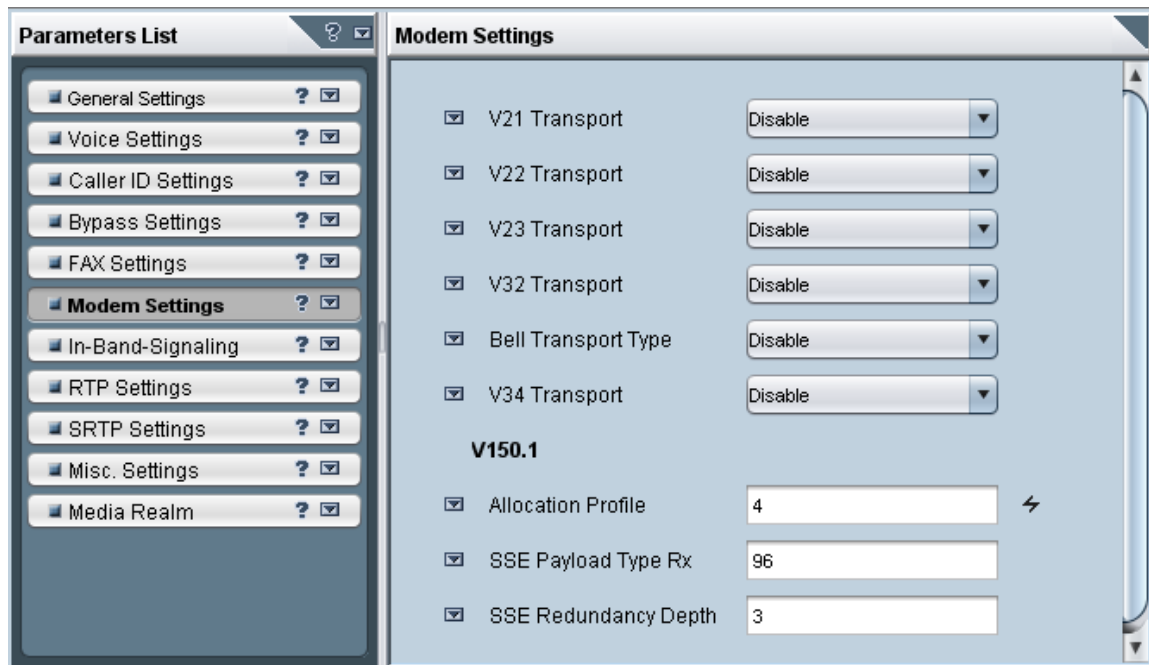


Figure 36: M3K Modem Settings for V.150 Screen

Set the Modulation Mode as follows:

- V32 Transport = 'enable'
- V34 Transport = 'enable'
- Allocation Profile = '11'
 - This controls how many DSPs on the TP6310 blade are to be allocated for V.150.1 Modem Relay capability.
 - Note: The default value of this parameter is "0", in which no modem channels are allocated. This must be changed for minimal operation.
- SSE Payload Type RX = "102" to be consistent with CM's preference.
 - This simplifies debugging if you get used to looking for a common value in the traces.
- SSE Redundancy Depth = 'Set to match Page 3 of CM's 'ip codec set' form.
 - Recommend to be set to '3.'

End of Application Note