

# IxLoad — Voice Test Solution

Validate that your voice service delivers a high quality of customer experience

## Problem: Maintaining Voice Quality in the Face of Exponential Traffic Increase

Voice delivery over convergent networks has its own challenges, as quality of experience (QoE) is strongly affected by latency and jitter. Measuring real-world call-flow characteristics to quantify how ever-changing traffic patterns affect voice is critical to the success of deploying voice communication solutions. Regardless if the solution is voice over LTE (VoLTE), unified communication (UC), call center, telepresence, or just regular voice over IP (VoIP), service providers and enterprises must meet consumer QoE expectations.

## Solution: Comprehensive Testing for Voice Delivery Platforms

Ixia's IxLoad is the industry's most scalable and flexible solution for realistic load testing of VoIP platforms to validate end-to-end voice delivery architectures. It emulates thousands of user sessions with dynamic call flows and measures real-time voice quality.

### Highlights

- Ensure voice delivery systems perform at peak loads and conform to interoperability standards
- Reduce post-deployment issues and customer churn by validating QoE using in-depth SIP and media analysis, including subjective quality of voice (POLQA / PESQ)
- Quickly detect and pinpoint performance issues by emulating complex, real world, multiplay multi-subscriber environments
- Reduce learning curve with library of pre-built test cases and intuitive drag-and-drop GUI
- Increase productivity with embedded intelligent state machine that automatically handles test details so test engineers can focus on the features under test

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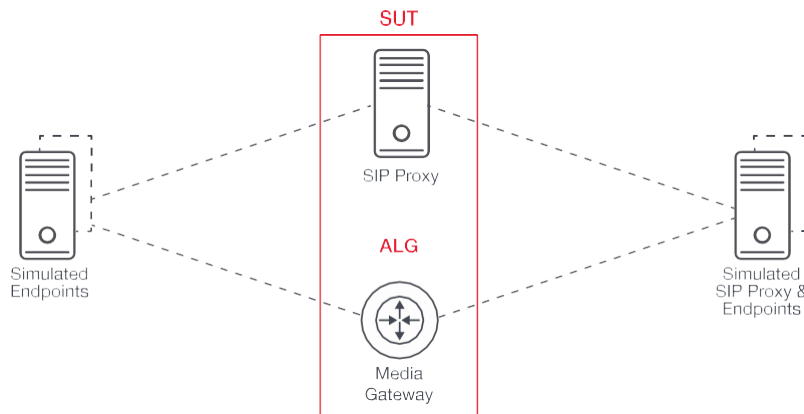
# Key Features

- Emulate real-world traffic using Ixia's highly scalable test platform or virtual ports
- Simulate detailed call control state machines, messages, and contents and create any test case, including negative testing
- Integrate with the real-time transport protocol (RTP) test library to generate voice, conversational video, and tones, and use a multitude of voice and video codecs to measure voice or video quality
- Fully automate feature and regression testing using the IxLoad TCL, REST, Python or Perl API

# Product Capabilities

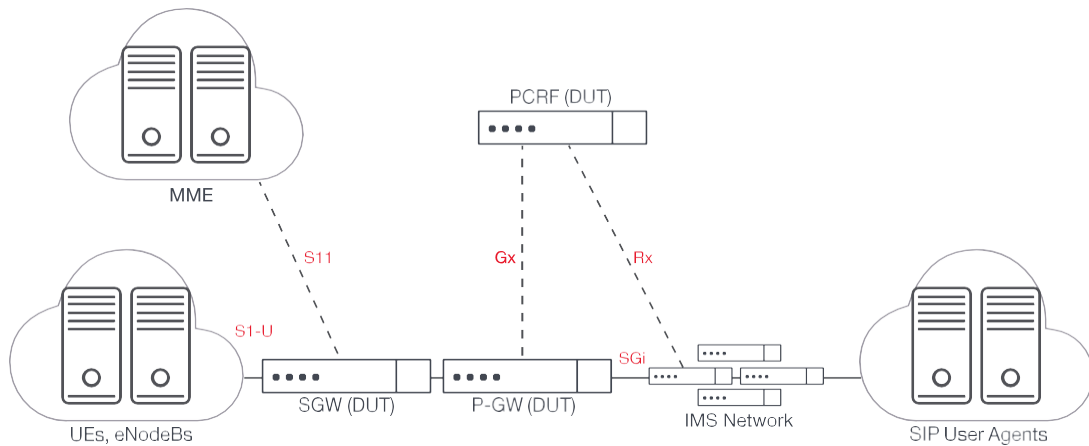
## Voice and conversational video quality assessment

IxLoad Voice is the perfect tool for functional, performance, and stability testing of SIP-based VoIP network components. Because IxLoad supports SIP, SDP, MGCP, H.248, SKINNY, and RTP protocols with voice codecs, video telephony, and fax in addition to video and data protocols, it can be used to test a variety of network components in VoIP, IMS, VoLTE and UC topologies.



**Figure 1.** IxLoad in a typical configuration simulating SIP endpoints and SIP proxy to test a distributed application layer gateway (ALG)

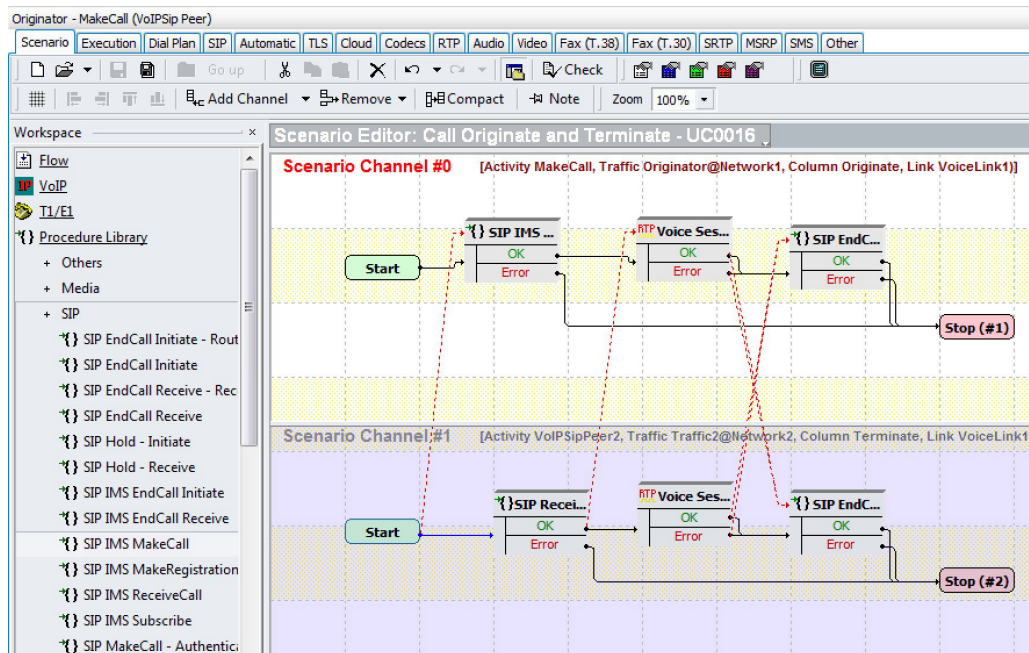
IxLoad provides IMS system and sub-system developers and providers with a complete solution to verify all aspect of IMS systems and their components. IxLoad is an invaluable tool for service providers and network equipment manufacturers (NEMs) for functional, protocol, conformance, regression and performance testing of end-to-end IMS networks and individual components. A single test can be used for both functional and load testing, on a single hardware platform.



**Figure 2.** IxLoad in a typical configuration simulating SIP user endpoint, MMEs, and eNodeBs for end-to-end LTE / IMS testing

## Scenario editor

The variety of implementations and protocol interpretations of VoIP, UC, IMS, and VoLTE solutions poses interoperability challenges not just for networks, but for testing as well. Test tools need to accommodate the variations and provide high flexibility in terms of the call flow (sequence of messages). IxLoad's Scenario Editor provides a powerful yet intuitive tool to edit the call flow and the content of control messages. It allows test engineers to address any interoperability issue and emulate any supplemental service for functional and negative testing. Innovative features allow users to easily create and modify test scenarios using a complete visual environment in which the technology-specific test functions and predefined test procedures are represented as visual blocks. Without any prior knowledge of programming or script languages, users can easily define tests by simple drag-and-drop operations.



**Figure 3.** DAQ samples physical parameters for capture and data analysis

The same test scenario can be used, without changes, at low scale or to emulate thousands or millions of endpoints or subscribers. This is achieved by the parameterization capabilities—the use of variables, expressions, and control functions—provided by the Scenario Editor.

## Event viewer

IxLoad provides a large set of statistics across metrics of interest for call control and media. However, the stochastic analysis of voice QoE it is not always sufficient. To identify and debug issues, it is not enough to know the percentage of the calls or subscribers with issues, it is important to know which call or subscriber failed. IxLoad's Event Viewer contains a great capability to pinpoint the exceptions and errors.

The screenshot shows the 'Event Viewer' window with three tabs: 'VoIP SIP Error Events', 'DI Errors', and 'All Events'. The 'VoIP SIP Error Events' tab is active, displaying a table with the following columns: Time, Chassis;Card;Port, Loop, Phone, Error..., Error Description..., Activity, and User ID. The table contains 8 rows of data, all showing 'SIP Exp...' errors with the description 'Can not evaluate ex...'. The 'Total Events' row at the bottom indicates 10 total events, 10 in view, and 0 dropped.

Time	Chassis;Card;Port	Loop	Phone	Error...	Error Description ...	Activity	User ID
2016/05/07 05:42:57	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	0
2016/05/07 05:42:58	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	1
2016/05/07 05:42:59	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	2
2016/05/07 05:43:00	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	3
2016/05/07 05:43:01	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	4
2016/05/07 05:43:02	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	5
2016/05/07 05:43:03	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	6
2016/05/07 05:43:04	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	7
2016/05/07 05:43:05	10.205.19.110;11;7	1	160000...	SIP Exp...	Can not evaluate ex...	VoIPsipPeer1	8
Total Events : 10		Events In View : 10		Events Dropped On Port : 0			

Figure 4. Event Viewer showing the errors for each emulated user endpoint

During a test, Event Viewer displays and logs error events per emulated endpoint, facilitating the diagnosis and correction of issues. By getting this information right away, test engineers can focus on addressing the issue instead spending time analyzing massive traffic captures.

## Specifications

### SIP

<b>IP Support</b>	IPv4 and IPv6
<b>Emulation</b>	SIP endpoints and user agents, SIP endpoints behind SIP proxies, SIP servers (proxy, registrar), IMS entities, trunks, gateways
<b>Features</b>	<ul style="list-style-type: none"> <li>• Maintains full control over SIP state machines, messages, and contents; creates any functional and load test case, including negative testing</li> <li>• Drag-and-drop GUI permits functional building blocks to be easily assembled into test cases and call flows with automatic protocol</li> <li>• rule enforcement</li> <li>• Session timers and message retransmission mechanisms are present with the option to be disabled by the user</li> <li>• When in SIP Proxy mode, the module routes the SIP messages based on content of the SIP messages, following the user defined rules</li> <li>• When in SIP Proxy mode, the library has the capability to interact with an HSS emulating the Cx interface</li> <li>• Supports WebRTC websockets</li> </ul>

## SIP

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	<ul style="list-style-type: none"><li>• Capability to test IMS devices and networks; x-CSCF isolation testing and emulation</li><li>• Includes IMS call profiles</li><li>• Graceful stop at ramp-down to end all the active calls at the end of test</li><li>• Test cases built for functional and feature testing can be reused for stress testing</li><li>• Emulates real-world traffic using Ixia's highly scalable test platform</li><li>• Simultaneously supports data, voice, and video protocols to emulate multiplay subscriber environments</li><li>• Tests a device's ability to sustain designed load levels or call attempt / connection rate</li><li>• Supports custom load profiles, which contain individual settings for each call mix element</li><li>• Performs call feature interoperability testing</li><li>• Provides ladder diagrams and media decoding with built-in packet capture and analyzer for in-depth SIP and RTP stream analysis</li><li>• Ships with library of pre-built test cases and call flows for easier startup</li></ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"><li>• Link layer protocols, including PPPoE, IPsec, and DHCP</li><li>• Transport layers: UDP, TCP, TLS (upto version 1.3), WebSocket, Secure WebSocket</li><li>• Diffserv type of service (TOS/DSCP)</li><li>• Real-world network configurations: multiple sub-networks, unique MAC addresses, 802.1q, 802.1p, and emulated router support</li><li>• VLAN tag with Q-in-Q support</li><li>• Configurable MAC addresses</li></ul>
<b>Commands</b>	<ul style="list-style-type: none"><li>• Atomic, flexible set of commands, to build any call flow:<ul style="list-style-type: none"><li>◦ Send Request, Send Response, Wait Request, Wait Response, Wait Message, Retransmit Last message, Extract Variables</li></ul></li><li>• Large set of predefined call flows procedures</li><li>• Integration with HTTP for IP Multimedia Evolution (IPME) use cases</li></ul>
<b>Methods</b>	<ul style="list-style-type: none"><li>• REGISTER, INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, SUBSCRIBE, REFER, PRACK, INFO, UPDATE, MESSAGE, User-defined methods</li></ul>
<b>Statistics</b>	Detailed user Endpoint, Proxy, Calls, and SIP statistics: <ul style="list-style-type: none"><li>• Number and rate of Calls, Active endpoints, Fail calls and endpoints</li><li>• SIP Call Times, SIP Delays, SIP Registrations,</li><li>• SIP Registration Rates, SIP Messages, SIP Errors</li><li>• VoIP/SIP Errors, SIP Busy Hour Call, SIP Cloud, SIP Other</li></ul>
<b>Media Capabilities</b>	<ul style="list-style-type: none"><li>• Integrated with the RTP test library to generate voice, DTMF, tones, and video; supports a multitude of voice codecs and the ability to test voice quality</li><li>• Supports Video Conference and Telepresence</li><li>• Emulates SIP endpoints submitting / receiving SMS</li><li>• Negotiates sessions with Fax over IP (T.38)</li><li>• Message Session Relay Protocol (MSRP)</li></ul>

## VOLTE

<b>Protocols Control plane</b>	<p>Supports stateful emulation of SIP and other Layer 7 protocols over eGTP to provide a comprehensive solution for VoLTE testing:</p> <ul style="list-style-type: none"><li>• 3GPP TS 29.274 Evolved General Packet Radio Service (GPRS) Tunneling Protocol for Control plane (GTPv2-C)</li><li>• 3GPP TS 29.281 GPRS Tunneling Protocol User Plane (GTPv1-U)</li><li>• IPv4 transport</li></ul>
<b>Protocols User plane</b>	<ul style="list-style-type: none"><li>• RTP, SRTP</li><li>• SMS, MSRP</li></ul>
<b>LTE Access Features</b>	<ul style="list-style-type: none"><li>• User equipment (UE) Category 1 – 4</li><li>• UE Category 5 under 2x2 MIMO</li><li>• FDD and TDD</li><li>• All 3GPP R8 2009 specifications</li><li>• 3GPP R9 June and December 2010 specifications</li><li>• All LTE FDD and TDD frequency bands</li><li>• All TDD configurations and SSF configurations</li><li>• Transmission modes; SISO, Tx Diversity, 2x2 MIMO and Beamforming</li><li>• 5, 10, 15, 20 MHz channel bandwidth support</li><li>• QPSK, 16QAM and 64QAM modulation schemes</li><li>• NAS compression and ciphering</li><li>• Full DL/UL HARQ capability</li><li>• Semi-persistent scheduling</li><li>• UE power control, group hopping</li><li>• Frequency hopping modes 0, 2, 4</li><li>• Automatic configuration of MIB/SIB parameters</li><li>• Support for default and dedicated bearers, with a maximum of 11 per UE</li><li>• UE initiated dedicated bearer creation, modification and deletion</li><li>• Network initiated dedicated bearer creation, modification</li><li>• Dynamic IP address allocation during session establishment</li><li>• Supports all LTE handover types</li><li>• Configuration of QoS and TFT per layer 7 activity</li></ul>
<b>GTP-c Features</b>	<ul style="list-style-type: none"><li>• Support for default and dedicated bearers, with a maximum of 11 per UE</li><li>• UE/MME initiated dedicated bearer creation, modification and deletion</li><li>• Network initiated dedicated bearer creation, modification</li><li>• Echo request and response</li><li>• Dynamic IP address allocation during session establishment</li><li>• Ability to configure the number of simulated MMEs and eNodeBs</li><li>• Supports eNodeB (X2) handovers with configurable event intervals and mobility paths between eNodeBs</li><li>• Supports S1-based handovers with indirect data forwarding tunnels</li><li>• Configuration of IMSI, MSISDN, IMEISV, RAC/LAC/TAC, MCC/MNC</li><li>• Configuration of QoS and TFT per layer 7 activity</li><li>• Prebuilt TFTs are supplied, with the ability to manually override with user TFT definition</li></ul>

## WebRTC

<b>Versions</b>	Long Polling WebSockets
<b>IP Support</b>	IPv4, IPv6
<b>Emulation</b>	WebRTC initiator or responder
<b>Features</b>	<ul style="list-style-type: none"> <li>• Emulates thousands of browsers interacting with an WebRTC server at unprecedented scale</li> <li>• Leverages the performance of Ixia’s scalable CloudStorm and PerfectStorm test platforms</li> <li>• Allows creation of customized media capabilities to exercise a wide range of emulated endpoints</li> <li>• Provides extensive real-time statistics including detailed signaling and media information</li> <li>• Simplifies call flow definition by using automatic handlers for initiated and responded requests</li> <li>• Offers flexible test configuration for emulated endpoint identity with support for user-defined Phone Book</li> <li>• Allows extensive validation of network capabilities with discreet control for endpoint solicited services at initial registration</li> <li>• Evaluates the user service experience using comprehensive QoE measurement and analysis for exchanged voice and video media</li> <li>• Gives immediate feedback on abnormal network reactions through test event notifications</li> <li>• Emulates the interaction between thousands of browser-over-RESTful APIs and the WebRTC server at massive-scale performance, allowing qualification of the desired platform for WebRTC deployment</li> <li>• Exercises the entire ecosystem behavior to validate end-to-end service delivery, leveraging Ixia’s CloudStorm and PerfectStorm test platforms to simulate massive-scale realistic traffic</li> <li>• Allows custom declaration of capabilities and a wider range of interoperability scenarios through configuration control of SDP negotiation parameters</li> <li>• Supports ICE</li> </ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"> <li>• Signaling transport layers: TCP, UDP, and TLS</li> <li>• Application signaling layer: HTTP and HTTPS</li> <li>• Real-world network configuration: multiple IP networks, unique MAC addresses, and VLAN tag support</li> <li>• Includes support for DTLS 1.1 and DTLS1.2</li> <li>• Includes support for STUN</li> </ul>
<b>Commands</b>	Make Call, Accept Call, End Call, Reject Call, Initiate Modify, Receive Modify, and Media Session
<b>Statistics</b>	<ul style="list-style-type: none"> <li>• Active Calls, Registration, Event Channel, Call Statistics</li> <li>• Latencies, Session Modifications, Errors, HTTP Requests</li> <li>• DTLS Handshakes, DTLS Latency, TCP, TCP Failures</li> <li>• RTP QoS, RTP Advanced QoS, RTP Jitter and Delay</li> </ul>

## WebRTC

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	<ul style="list-style-type: none"><li>• RTP Jitter Distributions, RTP MOS</li><li>• RTP MOS per Call Distribution</li><li>• RTP MOS Instant, SRTP</li><li>• RTP Consecutive Lost Datagram Distribution</li><li>• Video RTP QoS, Video SRTP</li></ul>
<b>Media Capabilities</b>	<ul style="list-style-type: none"><li>• Integrated with the RTP test library to generate voice, DTMF, tones and video. Supports a multitude of voice codecs and the ability to test voice quality</li><li>• Supports Video Telephony and Cisco Telepresence</li><li>• DTLS support for SRTP key negotiation</li></ul>

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## H.248 (MEGACO)

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<b>Version</b>	H.248 v1, v2 or v3 user selectable
<b>Transport</b>	UDP
<b>Emulation</b>	Media Gateways and Media Gateway Controllers
<b>Features</b>	<ul style="list-style-type: none"><li>• Maintains full control over H.248 state machines, messages, and contents. Allows the creation of any test case, including negative testing</li><li>• Drag and drop GUI permits functional building blocks to be easily assembled into test cases and call flows with automatic protocol rule enforcement</li><li>• Tests a device's ability to sustain designed load levels</li><li>• Supports custom load profiles, which contain individual settings for each call mix element</li><li>• Supports call feature testing under load</li><li>• Performs call feature interoperability testing</li><li>• Provides ladder diagrams and media decoding with built-in packet capture and analyzer for in-depth H.248 and RTP stream analysis</li><li>• Ships with library of pre-built test cases and call flows for easier startup</li></ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"><li>• Link layer protocols, including PPPoE, IPsec, and DHCP</li><li>• Diffserv type of service (TOS/DSCP)</li><li>• Real-world network configurations: multiple sub-networks, unique MAC addresses, 802.1q, 802.1p, and emulated router support</li><li>• VLAN tag with Q-in-Q support</li><li>• Configurable MAC addresses</li></ul>
<b>H.248 Profiles</b>	<ul style="list-style-type: none"><li>• "ETSI_ARGW" – ETSI H.248 Profile for controlling Access and Residential Gateways (NGN Release 1) / Reference doc: ETSI ES 283 002 V1.1.3 (2007-07)</li><li>• "ETSI_GateControl" – ETSI H.248 Profile for controlling gates between IP transport domains / Reference doc: ETSI TS 102 333 V1.2.0 (2008-01)</li><li>• "ETSI_BGF" – ETSI H.248 Profile for controlling Border Gateway Functions (BGF) in the Resource and Admission Control Subsystem (RACS) v1 / Reference doc: MSF-IA-MEGACO.009-FINAL</li></ul>

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## H.248 (MEGACO)

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	<ul style="list-style-type: none"> <li>• "ETSI_BGF" – ETSI H.248 Profile for controlling Border Gateway Functions (BGF) in the Resource and Admission Control Subsystem (RACS) v2 / Reference doc: ETSI ES 283 018 V2.3.0 (2008-05)</li> <li>• "ETSI_TGW" – ETSI H.248 Profile for controlling Trunking Media Gateways / Reference doc: ETSI ES 283 024 V1.1.4 (2007-07)</li> <li>• "MSF TGW" – MSF H.248 Profile for controlling an IP Trunking Gateway (Implementation Agreement Between a Call Agent and an IP Trunking Gateway) / Reference doc: MSF-IA-MEGACO.003.01-FINAL</li> <li>• "MSFR3TGW" – MSF H.248 Profile for controlling Trunking Media Gateways (Implementation Agreement for a MSFR3 MGC-2 Interface) / Reference doc: MSF-IA-MEGACO.011-FINAL</li> <li>• "MSF_BGF" – MSF H.248 Profile for Distributed Session Border Gateways (S-SBG/P-CSC to D-SBG Interface Implementation Agreement) / Reference doc: ETSI ES 283 018 V1.1.4 (2007-10)</li> <li>• "MSFUKAG" – MSF H.248 Profile for multi-service Access Gateway / Reference doc: MSF-IA-MEGACO.005-FINAL</li> </ul>
<b>H.248 Packages</b>	<ul style="list-style-type: none"> <li>• Basic</li> <li>• ETSI_ARGW/1</li> <li>• ETSI_GateControl/1</li> <li>• ETSI_TGW/1</li> <li>• MSF TGW/1</li> <li>• MSFR3TGW/1</li> <li>• ETSI_BGF/1</li> <li>• ETSI_BGF /2</li> <li>• MSF_BGF/1</li> <li>• MSFUKAG/1</li> </ul>
<b>Commands Simulated MGC</b>	<ul style="list-style-type: none"> <li>• Add, Modify, Subtract, Move, AuditCap, AuditVal</li> <li>• SrvChange(MGC), Wait SrvChange(MGC), Wait Notify</li> <li>• WaitRequests(MGC)</li> </ul>
<b>Commands Simulated MGW</b>	<ul style="list-style-type: none"> <li>• Wait Add, Wait Modify, Wait Subtract, Wait Move</li> <li>• Wait AuditCap, Wait AuditVal, SrvChange(MGW)</li> <li>• Wait SrvChange(MGW), Notify, WaitRequests(MGW)</li> </ul>
<b>Statistics</b>	<p>Detailed Media Gateway and Media Gateway Controller Statistics:</p> <ul style="list-style-type: none"> <li>• Number and rates of H.248 commands, transactions, retransmissions, protocol errors, received requests &amp; replies</li> </ul>
<b>Media Capabilities</b>	<ul style="list-style-type: none"> <li>• Integrated with the RTP test library to generate voice, DTMF, tones, and video. Supports a multitude of voice codecs and the ability to test voice and video quality</li> </ul>

## MGCP

<b>Version</b>	<ul style="list-style-type: none"> <li>• MGCP 0.1, MGCP 1.0</li> <li>• NCS 1.0, NCS 1.5 TGCP 1.0</li> </ul>
<b>Emulation</b>	Media Gateways and Call Agents
<b>Features</b>	<ul style="list-style-type: none"> <li>• Maintains full control over MGCP state machines, messages, and contents. Allows the creation of any test case, including negative testing</li> <li>• Permits easy assembly of functional building blocks into test cases and call flows with automatic protocol rule enforcement through a drag and drop GUI</li> <li>• Tests a device's ability to sustain designed load levels</li> <li>• Supports custom load profiles, which contain individual settings for each call mix element</li> <li>• Supports call feature testing under load</li> <li>• Performs call feature interoperability testing</li> <li>• Ships with library of pre-built test cases and call flows for easier startup</li> </ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"> <li>• Link layer protocols, including PPPoE, IPsec, and DHCP</li> <li>• Diffserv type of service (TOS/DSCP)</li> <li>• Real-world network configurations: multiple sub-networks, unique MAC addresses, 802.1q, 802.1p, and emulated router support</li> <li>• VLAN tag with Q-in-Q support</li> <li>• Configurable MAC addresses</li> </ul>
<b>Commands Simulated CA</b>	<ul style="list-style-type: none"> <li>• Send CRCX, Send DLCX (CA), Send MDCX, Send RQNT</li> <li>• Send AUERP, Send AUCX, Send EPCF, Wait NTFY</li> <li>• Wait DLCX (CA), Wait Command (CA), Wait RSIP</li> </ul>
<b>Commands Simulated MGW</b>	<ul style="list-style-type: none"> <li>• Send NTFY, Send DLCX (CA), Send RSIP, Wait CRCX</li> <li>• Wait DLCX (GW), Wait MDCX, Wait RQNT, Wait AUERP</li> <li>• Wait AUCX, Wait EPCF, Wait Command (GW)</li> </ul>
<b>Statistics</b>	<p>Detailed Media Gateway and Call Agent Statistics:</p> <ul style="list-style-type: none"> <li>• Number and rates of MGCP transactions, calls, commands, responses, messages, errors</li> <li>• Call times for CA and GW</li> </ul>
<b>Media Capabilities</b>	<ul style="list-style-type: none"> <li>• Integrated with the RTP test library to generate voice, DTMF, tones, and video. Supports a multitude of voice codecs and the ability to test voice and video quality</li> </ul>

## SCCP (SKINNY)

<b>Version</b>	Cisco SCCP version 5.0
<b>Emulation</b>	SCCP Phones
<b>Features</b>	<ul style="list-style-type: none"><li>• Assembles fully featured SCCP clients by simulating Cisco Skinny phones</li><li>• Supports call signaling: originate, answer, terminate calls, and advanced features</li><li>• Maintains full control over SCCP state machines, messages, and contents. Allows the creation of any test case, including negative testing</li><li>• Tests the call manager's ability to sustain designed load levels</li><li>• Supports custom load profiles, which contain individual settings for each call mix element</li><li>• Supports call feature testing under load</li><li>• Performs call feature interoperability testing</li><li>• Simulates device stations: 7902, 7910, 7935, 7940, 7960, 30 SP+, 12SP +, 12 SP, 12 S, 30 VIP, VGC</li><li>• Ships with library of pre-built test cases and call flows for easier startup</li></ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"><li>• Link layer protocols, including PPPoE, IPsec, and DHCP</li><li>• Diffserv type of service (TOS/DSCP)</li><li>• Real-world network configurations: multiple sub-networks, unique MAC addresses, 802.1q, 802.1p, and emulated router support</li><li>• VLAN tag with Q-in-Q support</li><li>• Configurable MAC addresses</li></ul>
<b>Commands</b>	<ul style="list-style-type: none"><li>• High Level Functions:<ul style="list-style-type: none"><li>◦ Register Client, UnRegister Client, Make Call, New Call</li><li>◦ Wait Call, Answer Call, End Call, Hold Call, On Hook</li><li>◦ Off Hook, Retrieve Call, Setup Transfer, Complete Transfer</li><li>◦ Remove Last Conference Party, Meet Me, Dial Digits</li><li>◦ Wait Digits, Transfer, Forward All Calls, Get Call Info</li></ul></li><li>• Low Level Functions:<ul style="list-style-type: none"><li>◦ Send Stimulus, Send Softkey, IsSoftKeyAvailable, Wait for Event</li></ul></li></ul>
<b>Statistics</b>	<ul style="list-style-type: none"><li>• Registration Rates, Registration Counters</li><li>• Registration Times, Busy Hour Call</li><li>• Call Rates, Call Counters, Call Time, Delays</li><li>• Errors, Channels, Loops</li></ul>
<b>Media Capabilities</b>	<ul style="list-style-type: none"><li>• Integrated with the RTP test library to generate voice, DTMF and tones. Supports a multitude of voice codecs and the ability to test voice quality</li></ul>

## Voice Quality Assessment

<b>IP Support</b>	IPv4 and IPv6
<b>Features</b>	<ul style="list-style-type: none"><li>• The RTP/SRTP engine and the voice and tones functions are integrated with all VoIP signaling modules</li><li>• Media capabilities are negotiated during the call setup phase using the automated generated or user defined SDP</li><li>• Maintains control of SDP state and generates media accordingly the SDP negotiated parameters</li><li>• Performs real time quality of service measurement using various metrics:<ul style="list-style-type: none"><li>◦ MOS R-Factor for audio,</li><li>◦ PESQ Perceptual Evaluation of Speech Quality</li><li>◦ POLQA Perceptual Objective Listening Quality Assessment</li></ul></li><li>• Plays real media clips and any user defined sequence of tones</li><li>• Detects and validate tone sequences</li><li>• Supports multiple type of media in same call and same test</li><li>• Record incoming audio streams and save decoded media as wave files</li></ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"><li>• Link layer protocols including: PPPoE, IPsec, DHCP, GTP</li><li>• Diffserv type of service (TOS/DSCP) for each type of traffic</li><li>• Real-world network configurations: multiple sub-networks<ul style="list-style-type: none"><li>◦ unique MAC addresses</li><li>◦ 802.1q</li><li>◦ 802.1p</li><li>◦ and emulated router support</li></ul></li><li>• VLAN tag with Q-in-Q support</li><li>• Configurable MAC addresses</li></ul>
<b>Commands</b>	<ul style="list-style-type: none"><li>• Voice Session, Talk, Listen, Multimedia Session</li><li>• Path Confirmation</li><li>• Generate DTMF, Detect DTMF</li><li>• Generate MF (Multi Frequency Tone), Detect MF</li><li>• Generate Tone, Wait for Tone</li><li>• RTP Control</li></ul>
<b>Audio Codecs</b>	<ul style="list-style-type: none"><li>• G.711 uLaw &amp; ALaw</li><li>• AMR-NB &amp; AMR-WB<ul style="list-style-type: none"><li>◦ Codec mode:<ul style="list-style-type: none"><li>- AMR-NB: 0 (4.75 Kbps) to 7 (12.20 Kbps)</li><li>- AMR-WB: 0 (6.660 Kbps) to 8 (23.85 Kbps)</li></ul></li><li>◦ Payload format: Octet aligned or Bandwidth efficient</li><li>◦ Mode Change capabilities</li></ul></li><li>• G.723<ul style="list-style-type: none"><li>◦ transmission rate: 5.3Kbps or 6.3Kbps</li></ul></li><li>• G.726<ul style="list-style-type: none"><li>◦ transmission rate: 16Kbps, 24Kbps, 32Kbps or 40Kbps</li></ul></li></ul>

## Voice Quality Assessment

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	<ul style="list-style-type: none"><li>• G.729<ul style="list-style-type: none"><li>◦ codec mode: A, B or A/B</li></ul></li><li>• EVRC</li><li>• EVS</li><li>• ILBC</li><li>• AAC-LD</li><li>• OPUS</li><li>• Custom codec<ul style="list-style-type: none"><li>◦ Imports and play back any RTP stream from a traffic capture file. It plays the RTP stream at the same rate, ptime, and with the same payload type as in capture file; any of these parameters can be also set by user</li></ul></li></ul>
<b>PESQ</b>	<ul style="list-style-type: none"><li>• Real-time Perceptual Evaluation of Speech Quality</li><li>• PESQ (P.862.1) and Active level (P.56) metrics</li><li>• Predefined balanced set of voice clips covering English's phonemes</li></ul>
<b>POLQA</b>	<ul style="list-style-type: none"><li>• Perceptual Objective Listening Quality Assessment</li><li>• ITU-T Rec. P.863</li><li>• Predefined balanced set of voice clips covering English's phonemes</li></ul>
<b>Tones</b>	<ul style="list-style-type: none"><li>• Real time Generation and Detection of DTMF (Dual Tone Multiple Frequency), MF (Multi-Frequency tones) or Single Tones sequences</li><li>• User configurable sequences</li><li>• User configurable MF and Single tones</li><li>• In band and out of band tone transmission</li><li>• 2833 Events and 2833 Tones</li></ul>
<b>RTCP</b>	<ul style="list-style-type: none"><li>• RTCP packets are sent automatically</li><li>• Sending the RTCP packets may be disabled by user</li></ul>
<b>SRTP</b>	<ul style="list-style-type: none"><li>• Conform with RFC3711</li><li>• Audio traffic over SRTP</li><li>• Crypto-suites:<ul style="list-style-type: none"><li>◦ NULL_128_NULL_32</li><li>◦ AES_CM_128_SHA1_32</li><li>◦ AES_CM_128_SHA1_80</li><li>◦ AES_CM_128_HMAC_SHA1_32</li><li>◦ AES_CM_128_HMAC_SHA1_80</li><li>◦ NULL_256_NULL_32</li><li>◦ AES_CM_256_SHA1_32</li><li>◦ AES_CM_256_SHA1_80</li><li>◦ AES_CM_256_HMAC_SHA1_32</li><li>◦ AES_CM_256_HMAC_SHA1_80</li></ul></li><li>• DTLS Support</li></ul>

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**Statistics**

- RTP Audio QoS, Delays, RTP Audio Advanced QoS
  - RTP Packet Errors, RTP DTMF, MF and Tone
  - RTP Audio MOS, Jitter Distribution
  - R-Factor and MOS degradation
  - RTP Consecutive Lost Datagram Distribution
  - RTP Playbacks and Records
-

## Voice Quality Assessment

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- Path Confirmation Errors, Path Confirmation Overall
  - Negotiated Codec Distribution
  - Negotiated Out Of Band Codec Distribution
  - Negotiated Packet Times Distribution
  - PESQ, PESQ distribution, P.56 measurements
  - POLQA scores
- 

## Conversational Video Quality Assessment

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<b>IP Support</b>	IPv4 and IPv6
<b>Features</b>	<ul style="list-style-type: none"> <li>• The RTP/SRTP engine and the conversational video functions are integrated with SIP - including VoLTE and WebRTC - H.248, MGCP and Skinny libraries;</li> <li>• Media capabilities are negotiated during the call setup phase using the automated generated or user defined SDP</li> <li>• Maintains control of SDP state and generates media accordingly the SDP negotiated parameters</li> <li>• Performs real time quality of video (VQMon)</li> <li>• Plays real media clips from an open pool of video clips</li> <li>• Supports Cisco Telepresence endpoint emulation</li> <li>• Support for Multiple stream per endpoint</li> <li>• Speaker rotation scheme for video conferencing testing:             <ul style="list-style-type: none"> <li>◦ All speak, Sequential, Random</li> </ul> </li> <li>• Supports multiple type of media in same call and same test</li> </ul>
<b>Network Capabilities</b>	<ul style="list-style-type: none"> <li>• Link layer protocols including: PPPoE, IPsec, DHCP, GTP</li> <li>• Diffserv type of service (TOS/DSCP) for each type of traffic</li> <li>• Real-world network configurations: multiple sub-networks             <ul style="list-style-type: none"> <li>◦ unique MAC addresses</li> <li>◦ 802.1q</li> <li>◦ 802.1p</li> <li>◦ and emulated router support</li> </ul> </li> <li>• VLAN tag with Q-in-Q support</li> <li>• Configurable MAC addresses</li> </ul>
<b>Commands</b>	<ul style="list-style-type: none"> <li>• Multimedia Session</li> <li>• RTP Control</li> </ul>
<b>Video Codecs</b>	<ul style="list-style-type: none"> <li>• H.263</li> <li>• H.264—AVC and SVC</li> <li>• VP8</li> </ul> <p>Video Codec Formats:            SQCIF, QCIF, QVGA, 525 SIF, CIF, 525 HHR, 625 HHR, VGA,            525 4SIF, 525 SD, 4CIF, 625 SD, SVGA, XGA, 720p HD, 4VGA, SXGA, 525 16SIF,            16CIF, 4SVGA, 1080 HD, 2Kx1K, 2Kx1080,            4XGA, 16VGA, 3616x1536 (2.35:1), 3672x1536 (2.39:1), 4Kx2K, 4096x2304 (16:9)</p>

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## Conversational Video Quality Assessment

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### Video Codec Profiles

Baseline (66)", Main (77), Extended, High Profile (HiP), High 10 (H10P), High 4:2:2 (H422P), High 4:4:4 (H444P), Scalable Baseline Profile

### Packetization modes of NAL Units:

Single NALU, STAP – A, Fragmented Unit – A

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### RTCP

- RTCP packets are sent automatically
  - Sending the RTCP packets may be disabled by user
- 

### SRTP

- Conform with RFC3711
  - Audio traffic over SRTP
  - Crypto-suites:
    - NULL\_128\_NULL\_32
    - AES\_CM\_128\_SHA1\_32
    - AES\_CM\_128\_SHA1\_80
    - AES\_CM\_128\_HMAC\_SHA1\_32
    - AES\_CM\_128\_HMAC\_SHA1\_80
  - DTLS Support
- 

### Statistics

- Video VQmon, Video QoS
  - Video Consecutive Lost Datagram Distribution
  - Video Playbacks
-

# Platform Options

visit [Keysight.com](https://www.keysight.com) for More Information on IxLoad Platform Options

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<b>Virtual Platform</b>	<ul style="list-style-type: none"><li>• IxLoad Virtual Edition (VE)</li></ul>
<b>Chassis</b>	<ul style="list-style-type: none"><li>• XGS-12 HSL/SD/SDL Chassis</li><li>• XGS-2 HSL/SD/SDL Chassis</li></ul>
<b>Load Modules</b>	<ul style="list-style-type: none"><li>• CloudStorm Fusion 10GE, 40GE, 25GE, &amp; 100GE</li><li>• CloudStorm 10GE, 40GE, 25GE, &amp; 100GE</li><li>• CloudStorm-XP Fusion 10GE, 40GE, 25GE, &amp; 100GE</li><li>• CloudStorm-XP 10GE, 40GE, 25GE, &amp; 100GE</li><li>• PerfectStorm Fusion 10/1GE, 40GE, &amp; 100GE</li><li>• PerfectStorm 10/1GE, 40GE, &amp; 100GE</li><li>• Novus-NP 10G/1G/100M</li><li>• Novus 10G/1G/100M</li><li>• Novus 10G/5G/2.5G/1G/100M</li></ul>
<b>Appliances</b>	<ul style="list-style-type: none"><li>• PerfectStorm ONE Fusion 10/1GE, 40GE, &amp; 100GE</li><li>• PerfectStorm ONE 10/1GE, 40GE &amp; 100GE</li><li>• Novus ONE Appliance</li><li>• Novus ONE Plus Appliance</li></ul>

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Note: Some of the IxLoad Voice features are available only on specific load modules

# Technology Solutions

Visit [Keysight.com](https://www.keysight.com) for More Information on IxLoad Technology Solutions

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- IxLoad Overview—Converged Multiplay Service Validation
  - IxLoad Virtual Edition (VE) L4-7 Application Performance Testing
  - IxLoad Data Test Solution
  - IxLoad Video Test Solution
  - IxLoad Voice Test Solution
  - IxLoad IPsec and Network Access Test Solution
  - IxLoad on AWS – Cloud Application Performance Testing
  - IxLoad on Azure – Cloud Application Performance Testing
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# IxLoad Ordering Information

## Chassis licenses

Part Number	Description
925-3371	<p><b>IxLoad Multiplay-Security-2016</b>, Software Bundle, Layer 4-7 Performance Test Application Data-Video-Voice-Security package. Includes:</p> <ul style="list-style-type: none"><li>• <b>Data:</b> Enables support for HTTP, HTTPS, TCP Session, FTP, DNS, Mail (SMTP, POP3 and IMAP), Database, SSH, RADIUS, TFTP, Application-Replay, DHCP, LDAP, Telnet, Stateless-Peer and StreamBlaster emulations.</li><li>• <b>Video:</b> Enables support for basic RTSP, IPTV (Multicast), Video-ADVANCED (VoD), Adobe Flash Client, Apple HLS Client, Microsoft Silverlight Client, Adobe HDS Client and DASH Client emulations. Includes Video Quality VQMON engine for up to 10Gbps and TCP VQ Video quality for TCP video traffic for up to 10 Gbps.</li><li>• <b>Voice:</b> Advanced VoIP SIP &amp; RTP, Audio Codecs, VoLTE extensions, and Bulk SIP &amp; MGCP. Includes: Voice Quality engine for up to 10Gbps, Video Quality engine for up to 10Gbps conversational video traffic.</li><li>• <b>ADVNET:</b> Enables support for Advanced Access networking protocols such as DHCP for IP address acquisition, DHCP Server, PPP, L2TP and IPsec.</li><li>• <b>Storage:</b> iSCSI, CIFSv1, CIFSv2 (SMB2), SMB3, NFSv3 Client, NFSv4 Client, NFS4.1 Client, Cloud Storage Client, DCBX, FCoE and FC; Note the FCoE and FC emulation is only supported on selected load modules.</li><li>• <b>Access:</b> DHCP, PPP, L2TP, L2TP, IPsec, IPsec Performance Benchmarking QuickTest &amp; SuiteB Cryptographic, 6RD, DSLITE and IPv6 SLAAC, 8021X, NAC and WEBAUTH.</li><li>• <b>Security:</b> Enables support for a collection of vulnerabilities and malware attacks with 1-year subscription service and DDoSv2.</li><li>• Software Impairment.</li></ul> <p>Note: some of the features are available only on specific load modules</p>
925-3341	<p><b>IxLoad Voice-ADV-2015</b>, Software Bundle, Layer 4-7 Performance Test Application; Enables comprehensive VoIP protocols, it includes Advanced VoIP SIP &amp; RTP, Audio Codecs, Video Conference, Video Codecs, Telepresence, MSRP, and VoLTE extensions. Also includes 925-3528 Voice Quality engine for up to 10Gbps, 925-3512 Video Quality engine for up to 10Gbps conversational video traffic and AVDNET-DHCP to emulate DHCP enabled clients. Also includes Skinny, H.248, Advanced MGCP, T.38 (Fax over IP), and Bulk SIP &amp; MGCP and Software Impairment on selected hardware.</p>
925-3372	<p><b>IxLoad Voice over LTE-2016</b>, Software Bundle, Layer 4-7 Performance Test Application. Enables VoIP protocols for VoLTE testing. Includes: Advanced SIP &amp; RTP, Audio Codecs, WebRTC, IPsec and VoLTE extensions. Includes: Voice Quality engine for up to 10Gbps, Video Quality engine for up to 10Gbps conversational video traffic, AVDNET-DHCP to emulate DHCP enabled clients and Software Impairment on selected hardware. Note: some of the features are available only on specific load modules.</p>
925-3422	<p><b>IXIA IXLOAD-PESQ</b>, node-locked perpetual license Optional Software; Enables real time PESQ measurement for VoIP. One license enables PESQ voice analysis on 300 channels. It runs on the controller of XGS-HSL or HGS-SDL chassis, on IxLoad VE, or on PerfectStorm ONE or Novus ONE appliances. TAA compliance is determined by hardware on which license is enabled.</p>
925-3423	<p><b>IXIA IXLOAD-POLQA</b>, node-locked perpetual license Optional Software; Enables real time POLQA measurement for VoIP; One license enables POLQA voice analysis on 12 channels. It runs on the controller of XGS-HSL or HGS-SDL chassis, on IxLoad VE, or on PerfectStorm ONE or Novus ON appliances. TAA compliance is determined by hardware on which license is enabled.</p>

## Appliance licenses

Part Number	Description
925-6321	<b>IxLoad Appliance Multiplay</b> , Software Bundle, Layer 4-7 Performance Test Application. Enables Data, Storage, Voice, Video and Access & VPN on PerfectStorm ONE and Novus ONE appliances. Includes: <ul style="list-style-type: none"><li>• 925-6121 IxLoad Appliance <b>DATA &amp; Storage</b></li><li>• 925-6112 IxLoad Appliance <b>VIDEO</b></li><li>• 925-6113 IxLoad Appliance <b>VOICE</b></li><li>• 925-6117 IxLoad Appliance <b>ACCESS &amp; VPN</b></li></ul>
925-6113	<b>IxLoad Appliance VOICE</b> , Software Bundle, Layer 4-7 Performance Test Application. Enables IxLoad Voice functionality on PerfectStorm ONE and Novus ONE appliances. Includes: Advanced SIP: SIP endpoint, Proxy and Cloud emulation TLS, SRTP Audio, Video Conference, Telepresence, MSRP, 925-3528 Voice Quality engine for up to 10Gbps, 925-3512 Video Quality engine for up to 10Gbps conversational video traffic

## Virtual edition licenses

Part Number	Description
939-9513	<b>IxLoad VE Tier-3 1G Floating SUBSCRIPTION</b> License. Includes the following IxLoad protocols supported on IxLoad VE for a duration of 1-Year: Data (HTTP, HTTPS, FTP, TFTP, DNS, DHCP, LDAP, Radius), Mail (IMAP, POP3, SMTP), Storage (SMB, NFS, iSCSI, Storage I/O), Voice (VoIP SIP, VoLTE), Video (DASH, Flash, HDS, HLS, IPTV VoD, MS IPTV, Silverlight), IPsec, IxLoad- Attack and IxLoad-AppLibrary. Enables 1 Gig throughput per unit.
939-9533	<b>IxLoad VE Tier-3 10G Floating SUBSCRIPTION</b> License. Includes the following IxLoad protocols supported on IxLoad VE for a duration of 1-Year: Data (HTTP, HTTPS, FTP, TFTP, DNS, DHCP, LDAP, Radius), Mail (IMAP, POP3, SMTP), Storage (SMB, NFS, iSCSI, Storage I/O), Voice (VoIP SIP, VoLTE), Video (DASH, Flash, HDS, HLS, IPTV VoD, MS IPTV, Silverlight), IPsec, IxLoad- Attack and IxLoad-AppLibrary. Enables 10 Gig throughput per unit.
939-9514	<b>IxLoad VE Tier-4 1G Floating SUBSCRIPTION</b> License. Includes the following IxLoad protocols supported on IxLoad VE for a duration of 1-Year: Data (HTTP, HTTPS, FTP, TFTP, DNS, DHCP, LDAP, Radius), Mail (IMAP, POP3, SMTP), Storage (SMB, NFS, iSCSI, Storage I/O), Voice (VoIP SIP, VoLTE), Video (DASH, Flash, HDS, HLS, IPTV VoD, MS IPTV, Silverlight), IPsec, IxLoad- Attack and IxLoad-AppLibrary. Includes EPC and WiFi Offload protocols. Enables 1 Gig throughput per unit.
939-9534	<b>IxLoad VE Tier-4 10G Floating SUBSCRIPTION</b> License. Includes the following IxLoad protocols supported on IxLoad VE for a duration of 1-Year: Data (HTTP, HTTPS, FTP, TFTP, DNS, DHCP, LDAP, Radius), Mail (IMAP, POP3, SMTP), Storage (SMB, NFS, iSCSI, Storage I/O), Voice (VoIP SIP, VoLTE), Video (DASH, Flash, HDS, HLS, IPTV VoD, MS IPTV, Silverlight), IPsec, IxLoad- Attack and IxLoad-AppLibrary. Includes EPC and Wi-Fi Offload protocols. Enables 10 Gig throughput per unit.

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