

Sangoma Technologies Corp.

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# **Netborder Express Gateway**

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Release Notes

Release 3.0.6

January 7, 2011

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# Netborder Express Gateway Release Notes

## 1 Product Compatibility

Here are some of the major compatibility points.

- Operating Systems Supported:
  - Microsoft Windows 2003 Server 32 bits
  - Microsoft Windows 2008 Server 32 & 64 bits
  - Microsoft Windows XP 32 bits
  - Microsoft Windows 7 32 bits
- Sangoma Telephony Cards Supported:
  - AFT A101/2/4/8 T1/E1 with hardware echo cancellation (PCI / PCI-Express)
  - AFT A200 Analog FXO/FXS with hardware echo cancellation (PCI / PCI-Express).
- Sangoma Software Release Versions Supported:
  - Driver : 6.0.30.0 (included in gateway software package)
  - Firmware
    - A101D : v37
    - A102D : v37
    - A104D : v37
    - A108D : v41
    - A200D : v13
- SIP 3261 compliant endpoints using either TCP, TLS or UDP as the transport protocol
- DTMF relay as per IETF RFC 2833.
- RTP/RTCP as per IETF RFC 3550/3551
- SRTP as per IETF RFC 3711
- Minimum Server requirements: Intel Core Duo 2 processor or later with a minimum 1 GB of RAM.

## Feature Support

Here is a brief list of the product features as supported in this release

<b>Feature</b>	<b>Notes</b>
<b><i>PSTN-initiated calling</i></b>	<p>Support FXO &amp; FXS analog interface</p> <p>Support ISDN-PRI Q931 (DMS100, 4ESS, 5ESS, National ISDN 2) terminal and network sides.</p> <p>NFAS for DMS100, 4ESS, 5ESS and National ISDN 2 variants terminal and network sides.</p> <p>NFAS with D-Channel backup is not supported</p> <p>BRI, CAS &amp; R2 signaling are not supported</p>
<b><i>SIP-initiated calling</i></b>	The Gateway listens on port 5066 by default.
<b><i>Support for 3xx redirect primitives</i></b>	Includes "hybrid" redirect (redirecting to either SIP or PSTN endpoint)
<b><i>SIP Registration</i></b>	Allows to register the gateway to a third party SIP registrar
<b><i>RTP processing as per RFC 3550 and RTCP as per RFC 3551</i></b>	G.711 codecs (uLaw and A-law) with law conversion.
<b><i>Secure RTP (SRTP) as per RFC 3711 and TLS v1.0 as per RFC3261 SIP/TLS requirements</i></b>	
<b><i>DTMF per RFC 2833</i></b>	Both DTMF relay (PSTN to SIP) and DTMF re-generation (SIP to PSTN)
<b><i>Mapping of PSTN calls to SIP endpoints through rules, including DNIS-based routing</i></b>	Configurable routing rules
<b><i>Mapping of SIP calls to PSTN ports, trunks and DN through rules</i></b>	Configurable routing rules
<b><i>CallerID/ANI/DNIS and custom information element</i></b>	Available in SIP message
<b><i>Packaged as a Windows Service</i></b>	
<b><i>Integrated into Windows Event Viewer</i></b>	
<b><i>Configurable logging per sub-system</i></b>	
<b><i>Call logs</i></b>	Per call information
<b><i>Web Service Interface for management</i></b>	
<b><i>SNMP</i></b>	DS00 & DS1 Mibs (RFC 2494 & 1232) and proprietary gateway call statistics MIB

## 2 Acquiring a License

The Gateway is licensed on a per telephony port basis. The license is host locked.

To obtain a **full multi-port license**, simply obtain the *MAC (Media Access Control)* address of the server and contact our support organization at 1800 388 2475 or direct at +1 905 474 1990 or email at [techdesk@sangoma.com](mailto:techdesk@sangoma.com).

To get the physical address of the Ethernet adapter, simply start a DOS command prompt and execute the following command: "ipconfig /all". Then look for the Physical Address item. It would look something like: 00-0B-DB-D8-06-00.

Please consult the user guide for more details.

## 3 Changes Since Last Release

### 3.1 Maintenance Release 3.0.6

- **New features**
  - This is a maintenance release.
- **Bug fixes**
  - Bug 5809: Comfort Noise Transmitter not resetting on Reinvite

### 3.2 Maintenance Release 3.0.5

- **New features**
  - This is a maintenance release.
- **Bug fixes**
  - Bug 5614: Gateway cannot be installed on systems with AMD 32-bit processors
  - Bug 5538: Unable to extract Caller Name from Facility or Display IE

### 3.3 Maintenance Release 3.0.4

- **New features**
  - This is a maintenance release.
- **Bug fixes**
  - Bug 5026: The description of the netborder.sip.contactHostName parameter is not accurate.
  - Bug 4996: CN packets noise levels can still be slightly out of the range.
  - Bug 5058: The first RTP packet, in the first call after a Gateway start is not marked.

### 3.4 Maintenance Release 3.0.3

- **New features**
  - This is a maintenance release.
- **Bug fixes**
  - Disabled Gateway Call stats MIB by default
  - Modified default SIP TLS/TCP transport mode to use synchronous I/O

- Modified default audio packet size between the application and the telephony boards from 10ms to 20ms
- Bug 5020: Restart gateway on failure check-box partially hidden when using custom CSS files.

### 3.5 Maintenance Release 3.0.2

- **New features**
  - This is a maintenance release.
- **Bug fixes**
  - Bug 4925 : Errors with gateway restart through quick setup wizard
  - Bug 4958: A lot of Gateway error messages in Event Viewer when taking FXS phone off hook
  - Bug 4986: Syntax errors in Gateway MIB files
  - Bug 4988: Changes to a standard MIB should be reverted or properly documented
  - Bug 4987: Syntax warnings in Gateway MIB files
  - Bug 4994: SNMP: dsx1ConfigTable -> dsx1TimeElapsed mib value should be hard coded to '0'

### 3.6 Maintenance Release 3.0.1

- **New features**
  - This is a maintenance release.
- **Bug fixes**
  - Bug 4257 : The gateway no longer create the C:\usr\snmp folder when the SNMP feature is enabled.
  - Bug 4896 : Removed hard-coded style on channel status page is preventing customers from applying a custom theme on that page.
  - Bug 4906 : Fixed performance issue that could be observed on some systems when the GwCallStats MIB is enabled
  - Bug 4915 : Fixed gateway SIP 'failover' mechanism when an INVITE cannot be sent to the primary host in TCP or TLS.
  - Bug 4935 : Modify parameters PSTN audio parameters to disable automatic level control by default and work with 10 ms audio packets between the application and the PSTN cards.

### 3.7 Release 3.0.0 General Availability

- **New features**
  - Windows 2008 64 bits

- VoIP
  - Support TLS as a SIP transport
  - Secure RTP (SRTP) support has been added to provide secure media transmission
  - The gateway can be configured to send a SIP 183 Session Progress immediately upon the reception of the SIP INVITE (Immediate 183). The purpose of that feature is to minimize the occurrence of media clipping (RFC 3960)
  - The gateway now responds to SIP OPTIONS requests sent for keep-alive purposes.
  - The gateway is able to handle numerous incoming RTP/SRTP streams in an outgoing SIP call that forks (RFC3960)
  - Silence detection and RTP Comfort Noise (CN) packet generation
  - Host name resolution to IP addresses can now be configured to bypass the host's mechanism and use DNS directly.
  - Configurable ISDN <--> SIP code mapping
- Management
  - SNMP support
    - DSO-MIB (RFC2493). The file \$INSTALLDIR/doc/mibs/DS0-MIB.txt contains the specification of that MIB.
    - DS1-MIB (RFC1232). The file \$INSTALLDIR/doc/mibs/DS1-MIB.txt contains the specification of that MIB.
    - Custom IF-MIB. The file \$INSTALLDIR/doc/mibs/SANGOMA-IF-MIB.txt contains the specification of that MIB.
    - Proprietary gateway call statistics MIB. The file \$INSTALLDIR/doc/mibs/GW-CALL-STATS-MIB.txt contains the specification of that MIB.
  - Gateway Manager
    - Users no longer need to manually enter the boards present in the server, they are detected automatically
    - Wanpipe id complexity is now hidden from users
    - The UI now allows the configuration of a failover SIP server for SIP outbound calls
    - Gateway Service auto start after reboot : can now be configured inside and outside the quick setup wizard
    - Gateway service restart after failure : can be configured inside and outside the quick setup wizard. Default is restart the gateway service in case of failure.
    - TLS & SRTP parameters have been exposed in the wizard.
    - PSTN outbound call configuration : users can now configure outbound dialing from the wizard without the need to edit routing rules.
    - The Web UI's quick setup wizard has been augmented to allow customers to perform custom post setup actions by running a script.
    - New Logging configuration panel
    - New Advanced properties configuration panel

- netborder-list-boards : a command-line tool to enumerate Sangoma boards present in the system
- netborder-properties-editor : a command-line tool to edit gateway or gateway manager properties files.
- SOAP API to control gateway call logging & production/development logging modes.
- Added new OAM command that can change global configuration parameters at run-time. Works only with gw.properties parameters having the runtime settable attributes set to TRUE. Examples:
  - netborder.gw.recordCalls.enabled
  - netborder.gw.recordCalls.directory
- Telephony
  - Added support of analog FXS telephony interface
  - International telephony tones
    - Tone generation configuration has been dramatically simplified. All countries are now supported.
    - Tone detection accuracy has been improved
- Audio quality
  - RTP Packet Loss Concealment (PLC)
  - Adaptive RTP Jitter Buffer
- Support interactive and silent installer modes
- **Bug fixes**
  - Bug 4099 : assertion during the gateway startup when a wanpipe number is false
  - Installer/uninstaller command line options are now documented (Bug 2847). Use the /H command line switch to get a popup with the information.
  - The Contact header sent by gateway sometimes does not have the correct SIP transport (Bugs 2853 & 2715)
  - Complete PRACK SIP message support
  - SIP fuzzing resilience : the gateway successfully passes all 4526 tests from the PROTOS test suite ([https://www.ee.oulu.fi/research/ouspg/PROTOS\\_Test\\_Suite\\_c07-sip](https://www.ee.oulu.fi/research/ouspg/PROTOS_Test_Suite_c07-sip) )
  - Fixed one way audio issue that may occurred after receiving T1/E1 error on A101, A102 and A104 boards.
  - Gateway may crash when it receives a SIP BYE just after its received SIP 200/OK for its own SIP BYE.
  - Echo canceler clock configuration on T1/E1 boards (bug 3890).
  - Fixed Analog caller ID detection for Canada lines.
  - Calling name not decoded with some switch in NI2 (bug 3717).
  - Garbage in received fsk caller-id and stelephony crash upon reception of long caller-id strings (bug 3364),
  - Manual level control is not supported for the the TO\_PSTN direction on A200 (bug 3339).

- Added British Telecom caller id detection support for analog FXO.
- Redundant events generated in the Windows event logs (bug 2876)
- May get assertion log while fast SIP REFER (bug 2915)
- Using wrong transport while sending SIP messages to the SIP user agent specified in the REFER message (bug 2922)
- Fixed Internet Explorer 8 and Firefox 3.5 support in the WebUI.
- Removed remaining gray column on the left-hand side of the browser when resizing the WebUI window.
- Fixed DTMF detection problem in E1.
- Installation application now allows to install the gateway in a system running on AMD processors configured in 32 bit mode.
- Bug 2684 – Gateway may send silence audio toward the PSTN interface when the time-stamp in the RTP stream makes jumps.
- Bug 2672 – Getting error message “pri\_xmt\_setup - buffer prepared exceed > max length” when sending ISDN SETUP message with large phone numbers.

### **3.8 Maintenance Release 2.1**

- New features
  - Servers using AMD processors are now supported
  - Windows 2008 Server 64 bits is now supported
  - Configurable ISDN <--> SIP code mapping
  - PRACK SIP message support in both directions
  - Interpret RTP Comfort noise packets to generate comfort noise towards the PSTN
  - ISDN messages can be written to a pcap file suitable for WireShark for troubleshooting purposes
- Bug fixes
  - Audio latency optimized in the SIP to PSTN direction
  - PSTN channels stay alarmed after a cable is reconnected in NFAS (bug 2564)
  - Gateway Manager Web UI sluggish in Chrome & Firefox on machines where IPV6 is enabled (bug 2622)

### **3.9 Release 2.0 General Availability**

- This software supports both analog FXO & digital PRI telephony interfaces.
- In band tone detection is now performed when required in PRI outbound calls.
- The SIP PRACK request is supported in calls incoming from the SIP interface.
- Added support of ISDN Network side (act like a telco switch) for all ISDN variants supported by the gateway.
- Added caller name support for all ISDN variants.
- Improved Web User Interface (UI)
- All gateway configuration files can be edited through the UI

### 3.10 Release 2.0 Limited Availability

- Extended of FXO connectivity to support most countries around the world (please consult the Web User Interface to get the list of these countries). However, the user has to edit tone definition files and the .RAM files used to regenerate call progress tones for all countries except AUSTRALIA, CANADA and USA. Consult the *Tone\_Configuration\_guide.pdf* for more details.
- Added support to set the Type Of Service (TOS) field in the IP header of the RTP and RTCP packets transmitted by the gateway via the parameter "*Netborder.media.ip.tos*" in the *gw.properties* file.
- Extended FXO disconnect supervision to support battery removal and reverse battery disconnect detectors.
- Improved audio quality.

### 3.11 Release 2.0 Beta

- This release is the first to offer FXO analog PSTN connectivity limited to North America countries.
- The Gateway Web User Interface has been redesigned and its capabilities have been greatly augmented :
  - The Gateway service can be started/stopped from the Web User Interface
  - Initial gateway configuration can be generated by a Web UI wizard
  - Most telephony configuration parameters can be modified through the Web UI.

### 3.12 Release 1.6.2

- Fixed problem where the Windows user interface was not responding to user input as soon as the user had started the gateway WEB interface on server where the gateway is running. This problem was observed only systems having a single core CPU.

### 3.13 Release 1.6.1

- Fixed a problem with the DTMF detection on B-Channel 23 of T1 spans.
- Fixed gateway crash that could occurs with some NFAS configurations.

### 3.14 Release 1.6.0

- Fixed an issue where the gateway was not able to receive or make outbound calls with some ISDN switches configured in National ISDN 2 (NI2). The problem was caused by unexpected information element contained in the RESTART ISDN messages produced by the gateway.
- Modified the default install directory.

### 3.15 Release 1.5.4

- Fixed an issue where the ISDN not able to establish the link when for ISDN group configured in NFAS with the PRIMARY span configured on the span different than the first span.
- Added a PSTN configuration ISDN parameter to disable or enable the initiation of the ISDN restart procedure when layer is coming up. This parameter (*initiateRestartProcedure*) is configurable per ISDN group. Please consult user guide for more details.

- Added new PSTN configuration ISDN parameters to control the behavior of the in-band progress tone generation (inBandProgressTonesGeneration) and the behavior of in-band progress tones indication (inBandProgressTonesIndicator). Please consult user guide for more details.
- Modified the gateway uninstall application to continue best- effort gateway removal process even if some errors occurs during the removal process. By doing this, the user can always uninstall the gateway.

### **3.16 Release 1.5.3**

- Fixed issue in E1 configuration where the ISDN layer 2 was not able to establish the link when the echo cancellation was enabled.

### **3.17 Release 1.5.2**

- Fixed RFC2833 multiple events for the same DTMF. Now consecutive events related to the same DTMF all use the same timestamp, as required by the RFC2833 specification. (Issue #1428)
- Fixed Error reporting on SIP message received with no 'Max Forward' header (Issue #1423).
- Fixed SIP messages SDP body so it ends with "\r\n". (Issue #1427)

### **3.18 Release 1.5.1**

- Fixed SIP stack behavior to process multiple 2xx responses as described in section 13.2.2.4 of RFC3261. The gateway establishes the session with the remote user agent described by the first received 2xx response and rejects (ACK and BYE) the sessions described by any other 2xx response received after that.
- Added a new configuration parameter to insert or not the "rport" parameter in the Via header of the SIP requests sent by the gateway. For more details, please consult the parameter "*Netborder.sip.includeRportParameterInViaHeader*" in the Appendix B of the user guide.

### **3.19 Release 1.5.0**

- Now support incoming SIP messages with Date header field with time zone offset +/- HHMM to accommodate third party user agents that are not compliant to the restrictions imposed by RFC-3261 with regard to the Date header field.
- Added support of NFAS for DMS100, 5ESS and National ISDN 2 variants.
- Modified the gateway installation program to pre-install Sangoma device drivers.
- Modified the SIP registration configuration file to specify the SIP transport to be used to reach the registrar and the registered contacts.

### **3.20 Release 1.1.3**

- Fixed parsing of the transport parameter contained in the URL of the Contact .header for all SIP messages.
- Now support sending 183 Session Progress on INVITE without SDP body.
- Added support of the Q.931 RESTART primitive.
- Fixed SIP to SIP redirect handling.

### 3.21 Release 1.1.2

- Fixed issue where the gateway failed to start because the gateway can't enable some Sangoma wanpipe devices after it reconfigure them. This behavior was observed on Windows 2003 server where there is one or more uninstalled third party devices.

### 3.22 Release 1.1.1

- Fixed issue where the gateway may loses one or more span over time. This issue was observed only when the gateway was running with 8 spans or more.
- Fixed issue where the gateway WEB user interface sometime failed to display channel statistics when the gateway is heavily loaded.
- Fixed various warning messages about unexpected events that could happen when the gateway is heavily loaded.

### 3.23 Release 1.1.0

- Added support for configurable voice quality enhancement features such as: acoustic echo cancellation, automatic level control, adaptive noise reduction and DTMF removal.
- Added support for a new echo cancellation mode optimized for speech recognizer. Also added configuration parameters for tail displacement, double talk behavior and comfort noise generation.
- Added support for E1 interfaces and ISDN-PRI Euro-ISDN (NET5, ETSI) variant.
- Fixed issue 1059: Added support of different number of physical interface and Isdn groups.
- Fixed issue 1013: Added configuration parameter "media.rtp.disableRtcp" to enable/disable RTCP on RTP port+1. When disabled, the port is still opened but received RTCP packets are dropped and none are sent.
- Fixed issue 955: Added support of B-Channel negotiation via the parameter "isdn/groups/group/BchannelNegotiation" in file pstn-config.xml.
- Fixed issue 946: Added support of OAM operations (Quiesce, In-service) on channels.
- Fixed issue 980: The message "ERROR - SangomaSpan::processDChannels> Sangoma::readMsg() failed (TIMEOUT)" does not appears in the logs anymore.
- Fixed issue 983: System performance does not degrade anymore when the gateway is restarted multiple times.

### 3.24 Release 1.0.2 (alpha)

- Modified the product name. This invalidates all licenses issued for the previous releases. Please send an email to support@sangoma.com to get a valid license for this new version of the gateway.
- Fixed issue 985: Added support of early media.
- Fixed issue 1042: Added support of SIP Re-invites.
- Fixed issue 1053: Added support of SIP REFER.
- Fixed issue 1055: Added support of call progress tone generation such as ring-back, busy, rorder and SIT tones when the gateway receive a SETUP message that is not end-to-end ISDN.
- Fixed issue 1061: The size the RTP packets (specified in file "gw.properties" parameters "Netborder.media.rtp.packetSizeMs") no longer need to follow the size of the Sangoma packets (specified in file "pstn-config.xml" for all sangoma interfaces via the attribute "voicePacketLengthInMs" of the BChannels sub element). However, voicePacketLengthInMs should be a common divisor of the all RTP packets. For example, set "voicePacketLengthInMs" to 10 ms to support 10, 20 and 30 ms RTP packets. If the gateway serves only 30ms RTP streams, set "voicePacketLengthInMs" to 30ms to get minimize the CPU usage and support more channels on the same workstation.
- Fixed issue 1067: The gateway reconfigures and restarts the WANPIPEs interfaces during the Gateway boot-up process.
- Fixed issue 1113: The gateway no longer crashes after several calls when it is connected to an Asterisk PBX.

### 3.25 Release 1.0.1 (alpha)

- Fixed issue 1067: The gateway is not limited anymore to RTP packets of 20 msec.

### 3.26 Release 1.0.0 (alpha)

- First version

## 4 Limitations and Known Problems

Here is the list of known problems and limitations.

### 4.1 Hardware & driver related limitations

- **AFT-400 analog telephony boards are NOT supported (bug 2235 & not tested).**
- **Support Sangoma Software version 6.0.30.0.** The gateway has been tested and validated with Sangoma software 6.0.30.0. The gateway validates this version and generates a warning if the version is different.
- **Echo cancellation tail length is fixed to 128ms for all calls.** The echo cancellation is performed by the hardware. Thus, having support for shorter tail length will have no impact of the overall performance of the gateway.

- Microsoft Windows Vista is NOT supported
- Bug 4126 : Analog drivers consume a lot of CPU
- Bug 4140 : T1 takes twice the time to come up with driver 6.0.30.0

## 4.2 Other limitations

- Bug 4186 : `getSystemUpTime()` can take up to 40ms to return. This affects the content of the DS1 Mib. All Mib fields that use the system up time are currently set to 0.
- Bug 4177 : When adding FXS interfaces in hardware detection mode through the PSTN Config tab in the WebUI, channel groups corresponding to the phone number of those interfaces are not automatically created. Hence, they have to be added manually for the gateway to be able to route calls to those interfaces.
- The gateway reports errors and fails to start when A200 boards with FXS modules are not connected to the power supply.
- Bug 3453 : DNS server list is interface dependent. The gateway does not send DNS requests to the right DNS server (primary) on a multi homed server when various interfaces are configured with different DNS servers. When all interfaces have the same DNS configuration, the right DNS server is used. There is a workaround for this bug.
- The gateway consumes 100% of the CPU when the server's clock is modified (Bug 3505).
- Analog PSTN interfaces seem to apply a gain on the output signal (bug 3272)
- Receiving a RE-INVITE does not force a rekeying in SRTP (bug 3242)
- International Telephony Tone Detection
  - Busy is erroneously detected as reorder or vice-versa in the following countries : Austria, Spain, Bulgaria, Denmark
  - Incorrect detection of several tones in the following countries : Jordan, Hungary, Philippines, South Korea, China
- Audio Quality. RTP Forward Error Correction (FEC) is not supported
- Wanpipemon tool
  - The gateway must be restarted after BERT tests are run for the gateway to be able to place calls on the tested channels.
  - The echo canceller must be disabled before BERT tests can be run
- **Documentation**
  - Included : Quick Start and Tone Configuration guides only
  - The Gateway User Guide is included in this distribution but is out-of-date, in particular the analog telephony support.
- **Analog disconnect supervision**
  - Low amplitude telephony tone are not always detected (Bug 1601)
- **The gateway does not support T.38 Fax relay.**
- **Gateway Web User Interface**
  - Supported web browsers : Internet Explorer 7+, Mozilla Firefox 3+.
- **FXO caller-ID**

- Support is limited to caller-ID extraction as described by Bellcore FSK 1200bps Caller-ID standards in SDMF or MDMF which is used in Australia, Canada, China, Hong Kong, New Zealand, Singapore and USA. The gateway extracts only the caller number from the caller-ID in SDMF mode and extract caller number and caller name in MDMF mode. ETSI FSK caller ID and caller name is also supported
- **ISDN RESTART procedure is not well handled (bug 2586).** In the seconds that follow a gateway restart, calls are abruptly terminated by the gateway.
- **Channels stay stuck in the Out-of-service state when we connect/disconnect a span in the out-of-service state. (Bug 2661)**
- **Servers with multiples NICs/IP addresses.** RTP traffic can only flow from a single, user-defined IP address (bug 1917).
- **Service shutdown while waiting to register/unregister to a SIP registrar may cause shutdown timeout:** If the feature of registering the gateway with a SIP registrar is used, and the gateway is waiting for a reply from a registrar that is particularly slow or down, it is possible that a service shutdown request times out in Windows before we can complete the operation (register or unregister). The impact is simply that the service shutdown is not very elegant.
- **Gateway Does Not Monitor the Via or Max hops Headers for Self-Loops:** If users design ill-formed routing rules, it could happen that they re-direct incoming SIP calls to the gateway's SIP user agent. The gateway does not currently ensure that the 'via' header is different from the source of the call nor that 'maxhops' is not violated. This could cause an infinite loop of SIP calls.
- **Limitations to the use of arbitrary SIP headers in the routing rules:**
  - If two headers of the same name are specified in the sip.out.header out parameters, only the last one is used
  - If a "known" SIP header (automatically generated by the gateway, as described in a point below) is used in sip.out.header, the header internally generated will not be overridden, creating two headers that have a great chance of confusing the remote SIP user agent.
  - Known SIP headers, automatically generated by the gateway, cannot be used as sip.in.header.\* parameters. The list of all known headers follows:

VIA,  
FROM,  
TO,  
CSEQ,  
CALLID,  
CONTENTLENGTH,  
ACCEPTENCODING,  
ACCEPT,  
ACCEPTLANGUAGE,  
ALERTINFO,  
ALLOW,  
ALLOVEVENTS,  
AUTHENTICATE,  
AUTHENTICATIONINFO,  
AUTHORIZATION,  
CALLINFO,  
CCDIVERSION,  
CONTACT,  
CONTENTDISPOSITION,  
CONTENTENCODING,  
CONTENTTYPE,

DATE,  
ENCRYPTION,  
ERRORINFO,  
EVENT,  
EXPIRES,  
HIDE,  
INREPLYTO,  
MAXFORWARDS,  
MIMEVERSION,  
MINEXPIRES,  
MINSE,  
ORGANIZATION,  
PRIORITY,  
PROXYAUTHENTICATE,  
PROXYAUTHORIZATION,  
PROXYREQUIRE,  
RACK,  
RSEQ,  
RECORDROUTE,  
REFERTO,  
REFERREDBY,  
REPLACES,  
REQUIRE,  
RESPONSEKEY,  
RETRYAFTER,  
ROUTE,  
SERVER,  
SESSIONEXPIRES,  
SESSION,  
SUBJECT,  
SUBSCRIBESTATE,  
SUPPORTED,  
TIMESTAMP,  
UNKNOWN,  
UNSUPPORTED,  
USERAGENT,  
WWWAUTHENTICATE,  
**WARNING.**