Sangoma Technologies Corp.

Netborder Express Gateway

Release Notes

Release 4.4.7

September 6, 2014

Netborder Express Gateway
Release Notes

1   Product Compatibility

Here are some of the major compatibility points.

- Operating Systems Supported:
    - Microsoft Windows XP 32 bits
    - Microsoft Windows 2003 Server 32 bits
    - Microsoft Windows 7 32 & 64 bits
    - Microsoft Windows Server 2008 32 & 64 bits
    - Microsoft Windows Server 2008 R2 32 & 64 bits
    - Microsoft Windows Server 2012
- Sangoma Telephony Cards Supported:
    - AFT A101/2/4/8 T1/E1/J1 with hardware echo cancellation (PCI/PCI-Express)
    - AFT A200 Analog FXO/FXS with hardware echo cancellation (PCI/PCI-Express)
    - AFT A400 Analog FXO/FXS with hardware echo cancellation (PCI/PCI-Express)
    - AFT A500/B500 ISDN-BRI with hardware echo cancellation (PCI/PCI-Express)
    - AFT B600 Analog FXO/FXS with hardware echo cancellation (PCI/PCI-Express)
    - AFT B700 Combo ISDN-BRI/Analog FXS with hardware echo cancellation (PCI/PCI-Express).
- Sangoma Software Release Versions Supported:
    - Driver : 7.0.10.0 (included in gateway software package)
    - Firmware
        - A101D : v40
        - A102D : v40
        - A104D : v45
        - A108D : v44
        - A116D : v44
        - A200D : v13
        - A400D : v11
        - A500D : v35
        - B500 : v43
        - B600 : v03
        - B700 : v35
- 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:
    - Less than 2 T1/E1s : Dual core processor (minimum 4MB cache) or better with a minimum 1 GB of RAM.
    - Up to 8 E1/T1: Quad core processor (minimum 6MB cache) or better with a minimum 2 GB of RAM.
    - Up to 16 E1/T1: Dual Quad core processor (minimum 8MB cache) or better with a minimum of 2GB of RAM

Feature Support

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

Feature	Notes
PSTN-initiated calling	Support FXO & FXS analog interface
	Support ISDN-PRI Q931 (DMS100, 4ESS, 5ESS, National ISDN 2, ISDN-Japan) terminal and network sides
	NFAS for DMS100, 4ESS, 5ESS and National ISDN 2 variants terminal and network sides
	NFAS with D-Channel backup is not supported
	Support ISDN-BRI terminal and network sides (point-to-point and multipoint)
	Advice of charge is not supported
	ISDN Overlap Dialing Reception supported
	ISDN-R2 signaling (Central and Southern America) terminal and network sides
	ISDN-CAS signaling is not supported
SIP-initiated calling	The Gateway listens on port 5066 by default
Support for 3xx redirect primitives	Includes "hybrid" redirect (redirecting to either SIP or PSTN endpoint)
SIP Registration	Allows to register the gateway to a third party SIP registrar
SIP HTTP Digest Authentication as per RFC2617 and RFC3261 section 22.4	Authenticate on INVITE and REGISTER requests only
TLS v1.0 as per RFC3261 SIP/TLS requirements	Authenticate and establish a secure link with SIP endpoints
RTP processing as per RFC 3550 and RTCP as per RFC 3551	G.711 codecs (uLaw and A-law) with law conversion
Secure RTP (SRTP) as per RFC 3711 and	Allow to exchange secure audio between endpoints
DTMF per RFC 2833	Both DTMF relay (PSTN to SIP) and DTMF re-generation (SIP to PSTN)
Fax relay as per ITU-T.38	Allows sending FAX across VoIP networks
	UDPTL transport only
	ITU-T V.34 is not supported
Mapping of PSTN calls to SIP endpoints through rules, including DNIS-based routing	Configurable routing rules

Mapping of SIP calls to PSTN ports, trunks and DN through rules	Configurable routing rules
CallerID/ANI/DNIS and custom information element	Available in SIP message
Packaged as a Windows	
FXO interface calibration	Calibrate the FXO interfaces in order to minimize the line echo
Integrated into Windows Event Viewer	
Configurable logging per sub-system	
Call logs and call recording	Per call information for diagnostic purposes
Web Service Interface for management	
Web User Interfaces for configuration and management	Allows to monitor and configure the gateway Contextual help
SNMP	DS0 & 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.

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.

3 Changes Since Last Release

3.1 Release 4.4.7 Patch Release

- **New Features**

  Added configurable keep alive timer for Bri links to prevent the line from going in power saving mode.

  Integrated Linux Drivers version 7.0.11.4.

3.2 Release 4.4.6 Patch Release

- **New Features**

  Integrated Linux Drivers version 7.0.11.0.

3.3 Release 4.4.5 Patch Release

- **New Features**

  Integrated Drivers version 7.0.10.0.

- **Fixed bugs**

  Bug R10353: Overlap Dialing: Remote T302 timer expiration not reported to upper layer

  Fixed some software crashing issues when handling malformed SRTCP packets.

3.4 Release 4.4.4 Patch Release

- **Fixed bugs**

  Bug R9978: Overlap Dialing: Allowed call to be routed when SENDING complete IE is received and no DNIS in the SETUP message.

3.5 Release 4.4.3 Patch Release

- **Fixed bugs**

  Bug R9471: Overlap Dialing: Expose T302 timer value configuration in PSTN config.

3.6 Release 4.4.2 Patch Release

- **Fixed bugs**

  Bug R7925: Overlap Dialing Support for PR

  Added support for Internet Explorer 11 web browsers

3.7 Release 4.4.1 Patch Release

- **Fixed bugs**

  Bug R8999: Channel Not Releasing

3.8 Release 4.4.0 Maintenance Release

- **New Features**

    Validated Lync 2013 Certification Compatibility.

3.9 Release 4.3.15 Maintenance Release

- **New Features**

    Integrated Drivers version 7.0.2.0.

    Added Support for digital A116 boards.

3.10 Release 4.3.14 Maintenance Release

- **New Features**

    Added support of overlap dialing reception for PRI channels.

    Added support for windows server 2012.

3.11 Release 4.3.13 Maintenance Release

- **New Features**

    Added Support of DTMF CallerID detection for analog FXO channels.

3.12 Release 4.3.12 Maintenance Release

- **New Features**

    Integrated windows driver version 6.0.46.0

3.13 Release 4.3.11 Maintenance Release

- **Fixed bugs**

    Bug R6671: Problem capturing information from Referred-By: header in INVITE from Lync

3.14 Release 4.3.10 Maintenance Release

- **Fixed bugs**

    Bug R7304: Patch to release call waiting SETUP messages for BRI channels

3.15 Release 4.3.9 Maintenance Release

- **Fixed bugs**

    Bug R6442: sip.in.referredBy* parameters are not extracted for INVITE requests

3.16 Release 4.3.8 Maintenance Release

- **Fixed bugs**

    Bug R6405: WARN - Unexpected eBOARD_CALL_ACCEPTED in state CALLENG_ACCEPTED

3.17 Release 4.3.7 Maintenance Release

- **Fixed bugs**

    Bug R6194: NBE Call Rejection Resources Unavailable.

3.18 Release 4.3.6 Maintenance Release

- **New Features**

    Integrated windows driver version 6.0.46.0

- **Fixed bugs**

    Bug R6230: Updated Norway Tones definition.

    Bug R5104: Mexico CAS R2 MF is not working in Mexico

3.19 Release 4.3.5 Maintenance Release

- **New Features**

    ◦ Added PSTN channel name to CDR logs.

3.20 Release 4.3.4 Maintenance Release

- **Fixed bugs**

    ◦ Bug R5021: Analog Call-Control slowing down real-time media threads under load.

    ◦ Bug 2539: Got ERROR SangomaAnalogBChannel::writeBChannelsDataSangoma::write() failed

    ◦ Bug R5014: Netborder light debian package uninstalls Wanpipe driver when updating

3.21 Release 4.3.3 Maintenance Release

- **Fixed bugs**

    ◦ Bug R4723: NBE system failure due to a race condition under heavy load while logging SIP messages.

    ◦ Bug R4817: ISDN Inband PROGRESS tone indicator selector in the WEB UI does not display the BOTH choice when configured in 4ESS or 5ESS.

3.22 Release 4.3.2 Maintenance Release

- **Fixed bugs**

    ◦ Bug R3813: wizard.properties file does not contains the proper variable to refer to installation directory.

    ◦ Bug R3793: Add configuration parameter to A108 card to control port numbering to match Sangoma DS3 mux numbering

3.23 Release 4.3.1 Maintenance Release

- **New features**

- ◦ Support for B500 BRI card
- ◦ New Windows wanpipe driver 6.0.45.1

- **Fixed bugs**

  - ◦ Bug R3483: Quick Setup Wizard - tab key not working correctly when FXO ports are installed.

3.24 Release 4.3.0 Maintenance Release

- **New features**

  - ◦ Support of ISDN-BRI Overlap Dialing reception for PSTN inbound calls.
  - ◦ New Windows wanpipe driver 6.0.45.0.
  - ◦ BRI span is now moved into an alarmed state when an outbound call on a channel, belonging to that span, fails because of a failure to bring layer 1 or 2 up (considered as cable unplugged).
  - ◦ Support package generation tool in the Web UI.
  - ◦ Support for UK caller Id on FXO before 1st without polarity reversal.

- **Fixed bugs**

  - ◦ Bug R3389 fixed : It is now possible to detect FSK caller-id even if there is not at least 500 ms of silence before the FSK data.
  - ◦ Bug R3393 fixed : BRI interface may some time fails to mount the physical layer.
  - ◦ Bug R3428: NBE BRI gateway does not ACK ISDN layer 3 RESTART messages.
  - ◦ Bug R3382: Generate a ring back tone when we get a 183.
  - ◦ Bug R3212: Improve BRI interface state management.

3.25 Release 4.2.2 Maintenance Release

- **Fixed bugs**

  - ◦ Bug R2972 – A200 FXO fails to dial out.

3.26 Release 4.2.1 Maintenance Release

- **New features**

  - ◦ Support of B500 BRI boards with hardware echo-canceller.
  - ◦ Support of A400 FXS/FXO boards with hardware echo-canceller.
  - ◦ Analog line status are now shown one line per board in order to save display space.
  - ◦ Character "," supported in phone numbers to pause dialling for 2 seconds (analog FXO - ref. 6942).

- **Fixed bugs**

  - ◦ Bug 6646 - IE fails to parse audio_form.js when opening Quick Setup tab.
  - ◦ Bug 6940 - Quick Setup lightbox does not work.
  - ◦ Bug 6947 - ISDN Cause Information Element is not decoded properly when the recommendation field is present.
  - ◦ Bug 6905 - SNMP gwCallDetailTable clears after every 15 minutes.
  - ◦ Bug 6958 - SIP port defaults to 5060 in TLS (instead of expected 5061).

- ◦ Bug 6613 - Decreasing "Failed to obtain call reference" message to DEBUG level as it only affects trace logging.

- ◦ Bug 6500 - R2 no local ringback generated towards PSTN on inbound calls - ringback gen and early media tested.

- ◦ Bug 6611 - Indicate in online help that BRI Point-To-Multi point is not supported in NETWORK mode.

- ◦ Bug 6526 - Report correct SNMP data for CAS/R2 calls + gwNumActivePstnInCalls and OutCalls SNMP call stats were inverted for all PSTN protocols.

- ◦ Bug 6598 - Import/Export and Provisioning configuration is out of date.

- ◦ Bug 6514 - FXO Calibration start page text clipping in lightbox.

- ◦ Bug 6462 - NBE should send CANCEL after SIP INVITE Timeout.

- ◦ Bug 6516 - DTMF digit duration has the wrong help text.

- ◦ Bug 6097 - Add lightbox page for Advanced configuration and Logging tabs.

- ◦ Bug 6645 - Add support for Firefox 5+.

- ◦ Bug 6580 - GW Dropping Call Park Music on Hold Media.

- ◦ Bug 6623 - IE 9.0 freezes in restart after confirming gateway restart.

- ◦ Bug 6569 - Missing end-line in log text.

- ◦ Bug 6654 - Importing configuration of the same version when version has a patch fails.

- ◦ Bug 6511 - Export fail because of an IndexError exception.

- ◦ Bug 6660 - lightbox_logging and lightbox_provisionning are broken.

- ◦ Bug R2181 - Don't disconnect call upon reception of progress with cause 127

- ◦ More robust detection of disconnect tones on long-lasting calls.

3.27 Release 4.2.0 Limited Availability Release

- • New Features

  - • Modified gateway manager configuration wizard to remember entered configuration parameters used to generate configuration files. When the wizard is run again, the wizard proposed the rememberred values in the wizard screens. The pstn-config.xml and routing-rules.xml files be overwritten according to the values specified while going through the wizard steps.

- • Configuration parameters changes

  - • Configuration wizard parameters are remembered in a new file $INSTALLDIR/config/wizard.properties.

  - • Added the following parameters in oamWebServer.proprerties:

    - • netborder.oamWebServer.wizardPropertiesFile

    - • netborder.oamWebServer.wizardParamDB

  - • Moved the following parameters from the oamWebServer.properties to wizard.properties. Refer to the following parameter mapping table.

oamWebServer.properties	wizard.properties

netborder.oamWebServer.defaultUDPListenPort	netborder.oamWebServer.sip.udp.listenPort
netborder.oamWebServer.defaultTCPListenPort	netborder.oamWebServer.sip.tcp.listenPort
netborder.oamWebServer.defaultTLSListenPort	netborder.oamWebServer.sip.tls.listenPort
netborder.oamWebServer.defaultUDPTCPSendPort	netborder.oamWebServer.sip.primarySipServerUDPTCPPort
netborder.oamWebServer.defaultTLSSendPort	netborder.oamWebServer.sip.primarySipServerUDPTCPPort

- Fixed bugs

    - Bug 6562 - Default audio law for BRI physical configurations should be A-law

    - Bug 6575 - Error when updating from 4.1.0 to 4.1.3 with BRI config

    - Bug 6577 - GW version is not displayed in the WEB UI

3.28

3.29 Release 4.1.3 Maintenance Release

- **New features**

    - Added support of ISDN-BRI multpoint

    - Added mechanism to configure gateway for a provisioning file generated by IPBX software such as 3CX

- **Know Limitations**

    - Status of the BRI channels remains in the IDLE state even if the cable is disconnected

    - Can't generate in-band progress tones (example: ring back) toward the PSTN for inbound CAS-R2 calls. Make sure the Telco switch is configured to generate progress tones.

    - Acoustic echo cancellation is not working properly. Please don't used it.

- **Configuration parameters changes**

    - PSTN configuration

        - Removed **TEI** ISDN-BRI configuration from the physical configuration

        - Added new ISDN-BRI call control configuration parameters: **topology**, **TEI** and **multipleSubscriberNumbers**.

- **Fixed bugs**

    - Bug 6489 – Can't detect reliably dial tone in some countries (example: Mexico).

    - Bug 6492 – Detection of the ringing signal coming on FXS is too long in some countries (example: Egypt)

    - Bug 6510 – Gateway may crash under certain conditions with ISDN-BRI cards.

    - Bug 6547 - High usage of SNMP & phone causes internal errors

    - Bug 6562 - Default audio law for BRI physical configurations should be A-law

    - Bug 6565 - Gateway does not always start with default license after update

3.30 Release 4.1.2 Maintenance Release

- **New feature**

  - Added support for Microsoft IE 9.0 and Firefox 4.0

  - Modified channel hunting algorithm to use digital channels prior to analog channels when the hunting is configured to REVERSE_LINEAR.

- **Know Limitations**

  - Bug 6489 – NBE does not reliably detect Mexico dial tone

  - Bug 6491 - INVITE with replaces scenario with no SDP in the invite seems to be broken

  - Bug 6500 -R2 no local ringback generated towards PSTN on inbound calls

- **Fixed bugs**

  - Bug 6443 - NBE SIP UAS bind on 0.0.0.0 for UDP transport even if we specify a single IP address

  - Bug 6462 - NBE should send CANCEL after SIP INVITE Timeout

  - Bug 6467 - NBE should support SDP "a=silenceSupp:on" for CN support

  - Bug 6468 -SNMP gwCallSource is invalid for J1 calls

  - Bug 6473: FXS/FXO bad audio when using 10ms audio packets between the application and the driver

  - Bug 6474 - Unreliable bringup on T1 lines, some enter alarmed state on gateway restart

  - Bug 6480 - Smart update issue with file paths

  - Bug 6483 – Mexico tone definition does not contains the SIT tone definition

  - Bug 6486 - GW is not processing presentationIndicator and screeningIndicator on restricted J1 calls.

  - Bug 6488 - Gateway manager start menu short cut is not working

  - Bug 6492 – FXO Ring Detection in Egypt

3.31 Release 4.1.1 Maintenance Release

- **Fixed bugs**

  - Fixed DTMF detector to avoid sending of RFC2833 events toward the SIP side due to residual echo present in the signal coming from the PSTN when a DTMF signal transmitted simultaneously toward PSTN.

3.32 Release 4.1.0 General Availability

- **New features**

  - Support to install above a previous version (3.0.0 and upper) and import existing configurations

  - Ability to export and import configuration via the WEB UI.

- **Know Limitations**

  - Bug 5345 – DTMF echo could be perceived when DTMF digits are transmitted toward an FXO interfaces even if the interface has been calibrated.

- **Fixed bugs**

  - Fixed various CAS R2 issues

  - Fixed various SIP protocol issues to pass more SIP torture tests

  - Bug 6048: FXO Calibration buttons clipped could be clipped when the WEB UI is used in light-box

  - Bug 6272: Quiesce-In Service is not working when the for 5ESS interfaces when the remote ends does not support SERVICE ISDN messages.

  - Bug 6283: SNMP Analog channels report dsx0Ds0ChannelNumber as 0 instead of 1.

  - Bug 6284: SNMP gwCallStats MIB: NoAnswer statistics is not counting unanswered calls properly

  - Bug 6285: SNMP gwCallStats MIB is not reporting the gwCallStartTime properly in the gwCallDetailsTable.

  - Bug 6424: RFC2388 events send by Linksys and Cisco phones are not interpreted properly because the marker bit is set on every packets.

3.33 Release 4.0.2 General Availability

- **New features**

  - Hardware

    - B600 Analog Voice Card – 4 ports FXO and 1 port FXS with hardware echo cancellation, PCI and PCI-Express

    - A500 ISDN BRI Voice Card – up to 8 ports with hardware echo cancellation, PCI and PCI-Express

    - B700 FlexBRI Hybrid Voice Card – up to 4 ports ISND-BRI and 2 Analog FXO or FXS ports with hardware echo cancellation, PCI and PCI-Express

  - PSTN

    - ISDN CAS R2 protocol (Central and South America countries only) terminal and network sides

    - ISDN-BRI terminal and network sides

    - FXO line calibration

  - SIP

    - HTTP Digest Authentication for INVITE and REGISTER methods

  - T.38 FAX relay

- **Know Limitations**

  - Bug 5264 : 401 and 407 messages are not recognized on invalid INVITE digest authentication

  - Bug 5268 : The gateway logs an ERROR at un-registration time when the last REGISTER was not answered

  - Bug 5449 : Bad behavior in Add/Remove boards in the PSTN configuration editor when removing a board and adding another if a FXS phone number is used in both boards.

  - Bug 5521 : AOC not supported in BRI

  - Bug 5642 : Call collisions (glare) not handled properly in R2

- Bug 5649 : R2 logs statement are not correlated to call-id

- Bug 5707 : In E1-R2, toggling from quiesce to in-service puts the gw in invalid state

- Bug 5720 : SNMP is not supported for E1-R2 and BRI interfaces

- Bug 5752 : R2 outbound calls failing with "handle_protocol_error" reported in SIP as 500 Internal Server Error

- **Fixed bugs**

  - Fixed various SIP interface issues to pass more SIP torture tests

3.34 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)

- 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 specifed 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 size 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.

3.35 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)

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3.36 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

3.37 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.

3.38 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 :

  o The Gateway service can be started/stopped from the Web User Interface

  o Initial gateway configuration can be generated by a Web UI wizard

  o Most telephony configuration parameters can be modified through the Web UI.

3.39 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.

3.40 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.

3.41 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.

3.42 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.

3.43 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.

3.44 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)

3.45 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.

3.46 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.

3.47 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.

3.48 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.

3.49 Release 1.1.1

- Fixed issue where the gateway may looses 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.

3.50 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 openned 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.

3.51 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 (specfied 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.

3.52 Release 1.0.1 (alpha)

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

3.53 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 7.0.10.0.** The gateway has been tested and validated with Sangoma software 7.0.10.0. 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

4.2   Other limitations

- Bug 5720 : SNMP is not supported for E1-R2 and BRI interfaces
- Bug 5264 : 401 and 407 messages are not recognized on invalid INVITE digest authentication
- Bug 5268 : The gateway logs an ERROR at un-registration time when the last REGISTER was not answered
- Bug 5521 : AOC not supported in BRI
- Bug 5449 : Bad behavior in Add/Remove boards in the PSTN configuration editor when removing a board and adding another if a FXS phone number is used in both boards.
- 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)
- **Gateway Web User Interface**
  - Supported web browsers : Internet Explorer, Mozilla Firefox.

- **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

- **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,
ALLOWEVENTS,
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.
```