Sangoma Technologies Inc.

NetBorder Call Analyzer

Release Notes

Version 2.0.8
April 21st 2015

NetBorder Call Analyzer - Release Notes

1 Product Compatibility

Here are some of the major compatibility points.

1.1 Standard Edition

- Hardware Requirements:
    - Quad-core CPU
    - 1 GB of RAM (2 GB recommended)
    - 60 GB of available disk space
- Operating Systems Supported:
    - Microsoft® Windows XP
    - Microsoft® Windows 2003 Server (32 or 64 bit version, but NOT IA-64)
    - Microsoft® Windows 2008 Server (32 or 64 bit version, but NOT IA-64)
    - RedHat® Entreprise Linux 5.x 64-bit (x86_64)
    - CentOS 5.x 64 bit (x86_64). Tested on CentOS 5.7
- Operating Systems **NOT** supported:
    - IA-64 version of the above operating systems.
    - All other operating systems
- SIP 3261 compliant endpoints using UDP or TCP as the transport protocol (TLS not supported)

2 Acquiring a License

NetBorder Call Analyzer is licensed on a per call analyzis port basis. The license is host locked. To obtain a **full license** (host-locked), obtain the **MAC (Media Access Control)** address of the system and use the Installation ID that came with the software to generate a license file. Please follow this URL:

- http://www.sangoma.com/support/register_netborder_software.html

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. On linux, the command is “ifconfig”.

Please consult the user guide for more details.

3 Limitations and Known Problems

Here is the list of known problems and limitations.

3.1 NetBorder Call Analyzer Engine limitations

- A call placed within the first 30 seconds of service start-up may fail due to initialization time of the application. (Ref. 3497)
- Silence suppression (VAD) is not supported during analysis period.
- Only G711 codecs are supported for the analysis phase. Other codecs may be used once the analysis is completed.
- Re-INVITE from called party is not supported until analysis is completed. (Ref. 3391)
- RTP/RTCP inactivity timers in the PSTN gateway/VoIP provider must be disabled if netborder.cpa.rtpProvider.type call-analyzer-engine property is set to packet-driven (Ref. 5009)
- SRTP/SRTCP transport for media is not supported.

3.2 NetBorder Call Analyzer Service limitations

- SIP REFER is not supported (Ref. 876)
- TLS transport for SIP not supported (Ref. 879)
- Reception of SIP 3XX Redirects not supported (Ref. 952)
- Sending of Reliable Provisional Responses following RFC 3262 is not supported (Ref. 1513)
- SIP REGISTER is not supported.

4 Changes Since Last Release

2.0.8

The following feature has been added for version 2.0.8 release:

- Added SIP INFO DTMF relay support when in connected state

2.0.7

The following feature has been added for version 2.0.7 release:

- Improved EAMD beep tone detection.
- Improved detection when no pre-connect audio is received

2.0.6

The following feature has been added for version 2.0.6 release:

- Bug #R10664 fixed: Core dumps generated when glare occurs during callee re-invite scenario.
- Bug #R10640 fixed: Reworked real-time worker tasks load balancing algorithms to avoid task overload issue.

2.0.5

The following feature has been added for version 2.0.5 release:

- Bug #R5473 fixed: Reverted the way tone detection is done to its original method.

2.0.4

The following feature has been added for version 2.0.4 release:

- Support of RTCP receiver report packets sending from call-analyzer-engine to callee (PSTN gateway/VoIP provider) during cpa sessions if netborder.cpa.rtpProvider.type call-analyzer-engine property is set to session-polling (Ref. 5009).

2.0.3

The following feature has been added for version 2.0.3 release:

- Relay of provisional responses (ref. 1863). *183 Session Progress* responses received from media gateways are now relayed immediately to dialer, minus the SDP description.
- Dynamically change pre-connect timeout on per call basis (re. 2359). The pre-connect timeout to use can be provided using a *RingTimeout* SIP header (value in seconds)
- T.38 FAX support (ref. 6909). Faxes relayed using the T.38 protocol are now detected in NCA.
- Configurable Codec list (ref. 6649). The preferred media codecs to announce when reaching the media gateway can now be configured in the NCA engine (*.rtp.encodingList* parameter)
- SIP OPTIONS requests can be relayed when using Genesys SIP Server with *oos-options-max-forwards* parameter. See *app.ForwardSipOptionsToRelayServer* configuration parameter in User's Guide.

The following limitations and problems were corrected:

- CANCEL requests do not appear in the logs (ref. 6449)

- Request URLs may be sent corrupted - using URL of simultaneous call under high call rates (ref. 6907)
- One call log created for each SIP OPTIONS request received (ref. 4000)

2.0.2

The following feature has been added for version 2.0.2 release:

- Per-call selection of Answering Machine detection mode via a prefix (Enh. 4932).
- Automatic provisional responses can be sent to stop dialer timers. (Enh. 4161 – parameter “app.nca.provisionalSentUponCallerInvite”)
- Robustness to Genesys SIP Server 7.6 error in *Content-Type* header value - “applicaiton/sdp” triggering “Illegal Sdp Negotiation“ error in NCA. (Enh. 4190)

2.0.1

The following feature has been added for version 2.0.1 release:

- Support of CentOS 5.x x86_64 and Red Hat Enterprise Linux 5.x x86_64

2.0.0

The following feature has been added for version 2.0.0 release:

- The detection of in-band telephony progress tones may now be performed in any country, based on configuration parameters and tones specification. Currently, more than 65 countries are pre-defined. Please refer to user guide for more information.
- The end of the greeting of an answering-machine may now be detected by NetBorder Call Analyzer to allow an application to leave a message at the appropriate time.
- New custom SIP headers are used to exchange more information with third party dialer applications.

The following limitations and problems were corrected:

- The various types of Special information tones (SIT) are all reported with a CPD-Result of 'Sit-Unknown'. (Ref. 3440)