WRTC 1.3R001 Release Notes

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About WRTC Release Notes

This page describes the enhancements done in Sonus WebRTC release V01.03.00R001.

Related Documentation

The Sonus WebRTC documentation is located at the following Sonus Networks Wiki space: WebRTC Services Solution Documentation.

Problems or Questions

For problems or questions, contact the Sonus Technical Assistance Center (TAC) via telephone, fax, or e-mail:

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About Sonus WebRTC Services Solution

Sonus WebRTC enables you to place a call to a contact center, participate in a multi-party audio and video conference, or engage in a screen sharing collaboration with colleagues over the internet or intranet without any additional plug-ins or downloads on your computer or mobile device. Any device that supports WebRTC enabled browser can be used to communicate with another WebRTC enabled browser or SIP call over the internet or intranet.

Interoperability

The WRTC interoperates with the following:

- SBC 5.1 to provide media service functionality when WebRTC endpoints are behind a NAT.
- EMS 10.0 with DM build [Insight EMS10.0.0-WRTC1.3.0-DM1.0.0] to register a WRTC node in to the appropriate cluster. The EMS also supports key functions including cluster configuration, faul and performance management.

Compatibility with Sonus Products

The WRTC 01.03.00R001 release is compatible with the following Sonus product releases versions:

Table 1: Compatibility with Sonus Products

WRTC V01.03.00R001	Supported Devices			
	SBC 7000 Series	SBC 5000 Series	SBC SWe	EMS
	V05.01.00R000	V05.01.00R000**	V05.01.00R000**	V10.00.00R000*

* WRTC 01.03.00R001 is explicitly tested with Insight EMS 10.00.00R000 by Sonus with DM build [Insight EMS10.0.0-WRTC1.3.0-DM1.0.0]

** Sonus tested with SBC 5200 and SBC SWe explicitly.

Obtaining an AMI Image

The following AMI is created for this release and shared privately with the customer AWS accounts.

Table 2: AMI IDs

Mode	Region	AMI ID
Appliance Mode	N.Virginia	ami-4164623a

AWS Appliance Mode Template Changes

The following updates are required for AWS templates:

- AMI ID change - Refer to **Obtaining an AMI Image** for new AMI ID.

For more information, refer to Appliance Mode Template File.

WRTC Compatible Browsers

WRTC SDK is explicitly tested on:

Table 3: WRTC Compatible Browsers

Browser	Version
Chrome	60.0.3112.101
Mozilla Firefox	54.0.1
Opera	42
IE	11

WRTC SDK is also expected to work on higher browser versions listed above.

Sonus WebRTC Installation Notes

For information on installation of WRTC refer to the Installing Sonus WebRTC Gateway.

WRTC EMA Login Credentials

To log on to WRTC EMA GUI, you must use the following credentials:

- Username - `admin`
- Password - `$0nus123`

Sonus API Update

The following new API `RTCPMux` is added in `Initializing WRTC SDK` section:

```
sonus.updateConfig("RTCPMux",value);
```

Table 4: API Parameter Description

Parameter	Description
RTCPMux	This is a boolean variable. • true - rtcpMuxPolicy is set to "require" in RTCPeerConnection object. • false - rtcpMuxPolicy is set to "negotiate" in RTCPeerConnection object.

This feature is tested with SBC 6.1R001 version and above.

Issues Resolved

This section lists resolved customer issues reported in previous releases.

Table 5: Issues Resolved

Issue	Type/Category	Description	Resolution
WSX-3434	Bug	200 OK for login is not sent back to SDK if the assigned number is carrying the country code.	The code has been modified to rectify an application error when the country code was attached in assigned number.
WSX-3548	Bug	Memory leak issues found in production network.	The code has been modified to resolve the memory leak issues during the error handling cases.
WSX-3606	Bug	Every time a new CDR file is generated for each record after the first files reaches the maximum file limit.	The code has been modified to resolve the error in calculation of maximum file size.
WSX-3762	Bug	RtcpMux support in WRTC SDK.	RtcpMux parameter is added in the config object based on which it is set to `require` or `negotiate` in the peer connection object.
WSX-3829	Bug	Implementing logout feature in WRTC EMA.	The logout feature has been implemented to resolve the issue.
WSX-3831	Bug	`wrtclogging.properties` file is packaged in DOS format by mistake.	The control characters are removed from the file to resolve the issue.
WSX-3833	Bug	Porting of WRTC 2.0 EMA features like user management.	The password change has been implemented to resolve the issue.

Known Issues

This section describes the known issues in this release.

Table 6: Known Issues

Issue ID	Category	Problem Description	Severity	Impact/Workaround
WSX-749	Application	Advanced call support in Firefox.	3	No Workaround
WSX-1378	CDR	Incorrect CDRs are generated when the conference invite is in ringing state.	3	Rename conference CDR's to "Conference Initiated" instead of "Conference Created".
WSX-1448	Application	File share request issue when group chat is upgraded to conference.	2	No Workaround
WSX-1563	Application	Unable to add two parties simultaneously in a P2P Conference.	3	No Workaround
WSX-1721	HA	Auto retrieval issue for three users in three node HA.	2	No Workaround

WSX-1720	HA	Three node HA issue for desktop request getting cleared after switchover, and unable to perform modality changes.	2	No Workaround
WSX-2107	HA	Unable to perform device transfer while switchover is going on in a WSX HA setup.	3	No Workaround
WSX-2706	Application	Number is assigned for users even when number profile is not attached to enterprise.	2	No Workaround

Known Limitations

Following are the known limitations in this release:

SIP Inter-working Limitations

- SIP Listener only listens to one port per transport type.
- Late Media is not supported.
- Support for SIP modality is only using `REINVITE` and not `UPDATE`.
- Upgrade from audio call to video call fails.
- Video call and hold from browser does not work as expected.
- Modalities fail in Firefox web browser, only audio and video calls, and mute and unmute features work in Firefox-to-SIP and SIP-to-Firefox communications.

P2P Conference

- During a conference call, only one user can initiate conference call hold and retrieve.
- Conference with SIP users is not support.
- Simultaneous file transfer is not supported.

Group Chat

- Group chat with SIP users is not supported.
- Simultaneous file transfer is not supported.

CAC Policy

- Multiple routes or prefix for the same user is not supported.

Multiple Point of Presence

- Logout and login to the peer device when hanging sessions appear in drop-down list.
- Device transfer and device pickup for SIP endpoints is not supported.
- The policy query is carried out only once per call. If the callee has MPOP with varying enterprise, the CAC policy is applied based on the POP chosen for first ring on the callee enterprise. Consecutive POPs will be rung if required, but no new CAC policy will be applied for different enterprises.

WRTC in Appliance Mode

- When client SDK delays in detecting a switchover, any signaling from the client within that time frame do not take effect.
- Call history is not supported.

WRTC EMA

- EMA stats displays node resource usage interval data at 5 minute interval. Rest of the stats are real time integrated stats.

Browser Media Statistics

- Collection of call statistics depends on web browsers such as Opera, Firefox, and Chrome. You must decide what to do in case the information is found to be incorrect or missing.
- WRTC SDK or WRTC Server does not take any action if the media parameters degrade.
- WRTC does not analyze statistics.
- Statistics are only collected at the end of each session. Any updates in the middle such as file transfer inside a call, upgrade audio call to video call, and again downgrade to audio may not be captured.
- Statistics are not provided by server in case of data channel.

Dryup

- Page refresh is required in case of disconnection with WRTC or load balancer after multiple dry-ups.

Statistics and Trap Related Limitations

- No statistics is generated for calls from SBC, if the destination number does not match the registered users.
- In case of switchover, statistics is not shared across the WRTC nodes.

Cluster Configuration

- Changes to keystore, trustore requires restarting WRTC application.
- Changes to port configuration requires restarting WRTC application.

EMS Interworking

- Web Application Policy profile is not used.

Multitenancy

- Unique number pool across enterprises must be configured.
- For multiple enterprises with multiple CAC policies, if CAC policies are different then inconsistent behavior is seen in case of multiparty (group chat or conference) scenarios with users from different enterprises.

Firefox

- RTC based DTMF is not supported in Firefox web browser. Only info based DTMF is supported.
- Re-offer/answer leads to failure of SetRemoteDescription in Firefox web browser (version 38 and later), with Chrome and SIP endpoints.

Generic SDK UC App

- The session does not exist when one of the users in a two party call, ends the call before the third user accepts the invite to join the call.
- When logged through ORTC, upgrading to a video call or downgrading to an audio call has a delay in media streaming by 6-7 seconds.
- In a group chat if users (A and B) upgrade to audio call and other users (C) upgrades to video, user C will not get the audio. This is because one way audio and other way video upgrade is not supported in icelink V2.
- Multiple upgrade requests on the same peer connection is not supported.
- Conference modalities are not supported.
- Group chat does not support file transfer.
- In a Web to SIP video call, call retrieval after hold clears the video call.
- ORTC/IE to SIP calls is not supported.
- Safari browser is not supported.