Release Note: MTA6328

Version: 4.2.93_1
Image sha1 digest: 846A0A1DB09ED158C7455807CB281EFBC1604DD5
File Size: 1,397,216 Bytes
Release Date: Jan 5, 2018

Bugs Fix:

1. CDR report wan-jitter use average interarrival jitter in ms.
2. Fixed RTCPXR Max Jitter Buffer Size. (It may less than nominal value before).

Historic Logs

############# 4.2.92_1 Notes ###########

Version: 4.2.92_1

Image sha1 digest: A0DB26054F66BF260B6E4FDB79BE8A33F2407837

File Size: 1,396,956 Bytes

Release Date: August 25, 2017

Bugs Fix:

1. Do not reset RTP statistics when jitter buffer resset. Only when ssrc changes, reset RTP statistics.
2. Correct parsing errors for some SIP headers in REFER messages.
3. Add SIP security features: Trust server list.
4. Update DNS information in the MTA BBS settings.

############# 4.2.91_1 Notes ###########

New Features

Bug Fixed

1. Patch the 3WayCall issue that remote holds the caller after being held. This issue was reported by customer. This is an unusual case of 3 way call where the caller presses flash hook and holds the remote, the remote also sends the invite to hold the caller.

Version: 4.2.90_2, Release Date: November 23, 2016###################

New Features/Enhancements

2. SIP INFO message support to allow for server-based 3-way Calling and Call Waiting using the 'application/broadsoft' Content Type. When flash hook server feature notify or info is enabled:
   - Allow flashhook during on hold
   - Process broadsoft type incoming info tune play

   Provisioning Tag:
   - SIPINFO_Flashhook_MIME_Type_1=application/broadsoft
   - SIP_INFO_FLASH=1
3. DNS SRV queries will be sent on the next registration (or when another SIP message is pending) after SRV TTL has expired. A record queries will be sent on the next

registration (or when another SIP message is pending) after A record TTL has expired.

  - If there is NO existing call, compare the current proxy IP with the list of A-record responses and, if the current proxy IP is on the list, stay with the current proxy; else switch to the 1st proxy IP on the list and do a REGISTER to the new proxy.
  - If there IS an existing call, cache the DNS A-record response and continue to re-REGISTER with the current proxy for the duration of the call. At the end of the call, compare the current proxy IP with the cached list of A-record responses and, if the current proxy IP is on the list, stay with the current proxy; else switch to the 1st proxy IP on the cached list and do a REGISTER to the new proxy (which will also trigger a new DNS query if TTL has expired)

4. Syslog server can be configured as fqdn and ip.

   Provisioning tag: SYSLOG_SERVER
5. Allow the following commands in CLI: RTCP-XR,force register de-register commands in CLI. Commands:
   - RTCP-XR: Cg
   - Force registration: Sn; Force de-registration: Sf
6. Dump the system configuration using command Ic.
7. Add PtimeSwitchEnable Setting to CLI: Cs.
8. Add EMS hearbeat type v4 and v5
9. Change wording DSCP to TOS.Add the following provisioning tags, TOS, TOS_VOICE,TOS_LAN, TOS_OTHER
10. Change telnet tcp timeout from 5mins to 15mins
11. When Ping to Host is enabled, a ping host can be configured to detect loss of uplink. The device pings the host every 15sec for a minute and then declares a network loss. If Ping to Host is disabled, the MTA triggers registration and upon registration failure to the sip proxy, the phone led turns amber and voip led turns off. The MTA also stops playing dial tone.

    Command: Ci->5->7

    Provisioning tag: PING_HOST_IP

Bug Fix

1. Fixed the issue of MTA reporting high packet loss numbers when rtp packets are out of sequence.
2. Fix prov tag Jitter_Buffer_Delay not working

############ 4.2.80_4 Notes ###########

New Features/Enhancements

1. Change the G726-32 to be dynamic payload codec.
2. Add heartbeat type V4 and V5 for EMS. Reporting MTA Registration Status. V4 is in plain status, and V5 is encrypted.
3. Add PTimeSwitchEnable setting to user configurable CLI console. Cs->c->42.

   The default value conforms to RFC3264. If the ptime attribute is present for a stream, it indicates the desired packetization interval that the offerer would like to receive.

############# 4.2.80_2 Notes ###########

New Features / Enhancements

1. Start RTP after 200 OK instead of ACK for incoming calls. To resolve voice issues happening in some long delay networks, e.g., satellite networks.

Bug Fix

1. Correct GUI display for QoS parameters. Changing DSCP to TOS.
2. Add provisioning tags: TOS, TOS_VOICE, TOS_LAN, TOS_OTHER

Known issue:

1. G.726-32 should use dynamic payload type.

############# 4.2.79_0 Notes ###########

New Features / Enhancements

1. Allow V21 preamble alone in both directions to trigger FAX mode.
2. Configurable SIP user agent header. (CLI command: Cs->c->26->e->x->c)
3. Updated BBS, optimizing FLASH writing.
4. Made polarity reversal and OSI exclusive after peer on-hook
5. Adding provisioning tag "Enable_Polarity_Reversal" and CLI "Cr" to control polarity reverse.

Bug Fix

1. Fixed MTA send out RTP when stream is receive only or inactive but with valid connection IP address.
2. If MTA is behind NAT of port forwarding enabled; however cannot be accessed externally.
3. Italian tone cadences
4. Tuned Tone playback level according to GR506.

############# 4.2.76_7 Notes ###########

1. T.38 settings now configured with the Ca command
2. Added feature in the Echo Cancellation – Non-Linear Cutoff Level – this is for devices with a high Gain level ie a phone that has a volume control set to a high value on some phones. Ct – 5 to configure this.

Added provision tag:

EchoCancel_HighCutoffLevel_#
1 - high cutoff level
0 - normal cutoff level

EchoCancel_TailLength_#
where TailLength can be 8ms to 32ms.

############# 4.2.75_2 Notes ###########

NEW DSP 2.6.10

1. Fix Call Waiting One Way voice
2. Fix image downloaded when using incorrect port for HTTP Provisioning
3. Fix additional M=audio being offered when you had G729 and G729A Codecs chosen.
4. DTMF Adjustment to address a phone issue of first digit not being detected because of RCA phone saturating tone when dial tone is present.
5. Improved Upgrade URL ip address lookup for image download.

Add iLBC Codec Support

For Provisioning Tag parameter for Codec. Now you can add iLBC as a choice. Ie, G711, iLBC, G729

Add Provisioning Tags –

DNDLine_X - Now either DNDLine_X or DNDLineX works

User_Name – Web Gui User Login Name

User_Password – Web Gui User Login Password

Prov_Server_IDs – Provisioning Server IP/FQDN

Prov_Server_Port – Provisioning Server Port

CAUTION: Make sure you have another provisioning Server up and running otherwise you will not be able to provision your devices until you bring it up, because you have changed where they look for the provisioning server with these two tags.

############# 4.2.72_2 Notes ###########

NEW DSP 2.6.6

1. Fixed VLAN to not tag items on LAN if no VLAN TAG is configured for it.
2. Fixed WEB GUI update of Preferred Codec – now takes effect without Reboot
3. Fixed Ga (Voice Gain) command to allow values greater then 10
4. Add SNTP in WEB GUI
5. Add Provisioning Tag of FQDN – Provision a hostname into the unit – equivalent to the Mn -> 3 CLI command.

Know Issues:

Blind Call transfer with *90: The transferor can't play busy tone after using *90 transferring this call to transferee.

Consult Call transfer with *90: The transferor plays dial tone instead of busy tone after using *90 transferring this call to transferee.

############# 4.2.70_2 Notes ###########

NEW DSP 2.5.35

1. Fixed a DTMF Playback error in Conferencing
2. Fixed RFC2833 playback error in G.723
3. Fix port 3 Ring in 8 port device

4. Fix Restore to default issue of Loop/Ground start giving random values, all will be Loop Start now.

############# 4.2.69_36 Notes ###########

New DSP 2.5.33 Tue Aug 09 11:32:47 2011

1. Fixed line EC update buffer pointer problem.
2. DTMF detection / playback duration regulation.
3. Fixed G.728 decoder gain error.
4. Loop back error in silence suppression
5. RFC2833 DTMF playback error in conferencing

1. change "nc=" value to lowercase hex.
2. Fixed fax/modem connection issue:
    a. jitter buffer adaptive mode error in fax/modem mode.
    b. malformed rtp packets during codec changing
    c. network side bell answer tone detection (dsp)
    d. some fax/modem tone dropped
    e. end of fax/modem tone causing delay buffer clean up.
3. Modified current jitter buffer status display (Tj)
4. Fixed - syslog msg has wrong CallerID & Callee ID
5. In Alert-Info process: Added BellcoredrX control for callwaiting tone."
6. SDP Media Match Factory Default setting changed to disabled
7. When DND is enabled, user will hear a indication tone before dial tone.
8. Enabled per session RTCP-XR
9. Fixed - reboot needed if Tx and Rx gain are changed via MTA web page.
10. Fixed - Ptime allows you to enter a value above the Max
11. Fix telnet crash problem

Added Features

1. Add DONOT_DISTURB_ENHANCE to generic
2. Add WEB GUI Access Control for the Router/Switch Mode and WEB. LAN, SNMP access to generic.

Add the following provisioning tags.

1. PTimeSwitchEnable
2. StaggeredRingEnable
3. DNDOnDS: invoke string for enabling DND
4. DNDOffDS: invoke string for disabling DND
5. DNDFeature: 1 to enable DND feature,0 to disable DND
6. DNDLine[n]: 1 to enable DND on port n, 0 to disable DND on port n note: n starts from
7. ENABLE_RTCP_XR for enable/disable rtcp-xr