

New Features Guide

MAXCS 7.5

October 28, 2014

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Introduction

This guide describes the enhancements that have been added in MaxCS Release 7.5. These features are fully documented in the MAXCS 7.5 documentation. Related documentation includes:

- *MAXCS 7.5 Upgrade Guidelines* – Describes how to upgrade to release 7.5 from older versions of MAXCS
- *MAXCS 7.5 Polycom Configuration Guide* – Explains how to configure Polycom phones for release 7.5
- *Polycom User Guide* – Shows users how to perform common tasks with their Polycom phone

Contact your AltiGen representative to receive copies of these documents.

Operating System Support

This MAXCS 7.5 release supports Softswitch and MAXCS Private Cloud ONLY. Hardware chassis support will be added in a later update.

The following operating systems are supported:

- Windows 2008 R2 64-bit SP1
- Windows 2012 Server R2 64-bit
 - Supported on VMWare ESX 5.5 only
- Windows 7 Professional 64-bit SP1
- Hyper-V version 6.1 and VMWare ESX 5.5

Release 7.5 requires JRE 1.8; therefore, MAXCS 7.5 does not support Windows 2003.

Enhancements Included in Release 7.5

The MAXCS 7.5 release includes enhancements in the following areas:

- Polycom Enhancements – These features are covered in the MAXCS 7.5 *Polycom Configuration Guide*.
 - Easier Polycom deployment, centralized administration (Hours Offset, Time Format, Digit Map, TLS and SRTP support), 3rd Party call control, #17 Extension relocation for the same extension
 - New Polycom Advanced Features license unlocks additional functionality: BLF, Line Park, DND synchronization with clients, Polycom group paging (Polycom to Polycom only, broadcast limited to local LAN), Client Auto Answer (Polycom VVX models only)
- Licenses
 - Sightmax seat license
 - Polycom Advanced Features license
- Codec Enhancements
 - G.722 Codec Support
 - G.722 Supported Devices/Interfaces
 - G.722 License Information
 - G.722 in HMCP Board Configuration
 - G.722 in the Current Resource Statistics View

- Managing Codec Priorities
- Managing IP Network Bandwidth
- Codec Negotiation
- Codec Upgrades
- Codec Preference for Incoming SIP Trunk Calls
- Codec Preference for Outgoing SIP Trunk Calls
- Media Enhancements
 - Media pass-through (HMCP)
 - T.38 pass-through
 - FoIP extension
 - FoIP Hunt group
 - Fax routing
- Enhanced SNMP trap monitoring
 - SIP Trunk group status
 - Out of license condition
 - Out of Codec/SIP Trunk channels conditions
 - Abnormal conditions (911, suspicious frauds, malicious SIP)
- HMCP tool enhancements
 - DPC Latency measurement
 - Generating Disk IP Load
 - Generating CPU Load
 - G.722 Codec Load
- Additional enhancements, which include a new *repeating recording* tone for workgroup call recording, a checkbox for blocking unauthorized SIP invite requests, and a SightMax version field – See [Additional Enhancements](#) starting on page 19.
- Client enhancements
 - Handling client calls via some models of Polycom phones (requires an Advanced license)
 - Client auto-answer with some models of Polycom phones (requires an Advanced license)
 - Synchronize only Private Contact folders

Features No Longer Supported in Release 7.5

As of Release 7.5, MAXCS no longer supports the following features:

- ACC is no longer supported
- Windows 2003 Server OS for Softswitch is no longer supported
- TAPI is no longer supported
- H.323 SIP Tie-Trunk is no longer supported



Important: Any IP Dialing Table entries that have *H.323* set for the Protocol field must be changed to *SIP* before you upgrade to release 7.5

- The *TLS enabled* and *SRTP disabled* settings are no longer supported

- The Persistent TLS connection for SIP tie-trunk is no longer supported
- Redundancy is no longer supported
- Multiple gateway Softswitch is not supported

Additional Changes

Please note the following additional changes in Release 7.5:

- AltiGen Phone Group Paging uses a SIP Tie Trunk channel in this release.
- If Polycom phones are used, you must select **RFC 2833** for the *DTMF Transport* field in the Codec Profile assigned to the SIP tie-trunk. Also, the Release-Link-Tie over SIP tie-trunk feature is not supported in this configuration.
- The provisioning process for Polycom phones is different in Release 7.5. Refer to the *MaxCS 7.5 Polycom Phone Configuration Guide* for phone setup procedures.

Enhanced Polycom IP Phone Features

This release of MaxCS includes many additional features for various Polycom phones. Some of these features require a new MaxCS license, the *Polycom Advanced Features* license.

Basic Polycom Enhancements

The following enhancement apply to all supported Polycom phone models.

- Easier configuration and deployment
- Centralized administration (Hours Offset, Time Format, Digit Map, TLS and SRTP support)
- 3rd Party call control
- Station login – users can log into their extension on a different phone

Advanced Features

The new *Polycom Advanced Features* license unlocks additional functionality.

- BLF slots for line monitoring and call pickup
- Line Park slots
- DND sychronication with MAXCS clients
- Polycom group paging (Polycom to Polycom only, broadcast limited to local LAN)
- Client Auto Answer (Polycom VVX models only)

For full configuration instructions, refer to the *MaxCS 7.5 Polycom Configuration Guide*.

New Licenses

Several new licenses are offered for MAXCS 7.5:

- **Polycom Advanced Features license** (ALTI-SIPDEVADDON-PLCM-XX) – Refer to the *Polycom Configuration Guide* for lists of the specific features that are supported by each Polycom model.

- **SightMax Seat license** (MAXCS-SIGHTMAX-xx) – This license supports integration with SightMax Enterprise Edition software (for instructions on configuring and using the MaxCS SightMax Integration service, refer to the *MAXCS 7.5 SightMax Configuration Guide* (when available).

Codec Enhancements

Codec enhancements in Release 7.5 include the following features:

- G.722 Supported Devices/Interfaces
- G.722 License Information
- G.722 in HMCP Board Configuration
- G.722 in the Current Resource Statistics View
- Managing Codec Priorities
- Managing IP Network Bandwidth
- Codec Negotiation
- Codec Upgrades
- Codec Preference for Incoming SIP Trunk Calls
- Codec Preference for Outgoing SIP Trunk Calls

MAXCS 7.5 supports the G.722 codec (for Softswitch and Cloud deployments; G.722 is not supported on hardware chassis deployments).

G.722 is a wideband audio codec that provides higher voice quality than more narrow-bandwidth codecs such as G.711. MAXCS supports G.722 64 kbps only.

The CPU processing requirement for G.722 is similar to G.723 and G.729. It is designed to use the same combo codec resources as G.723 and G.729.

Release 7.5 offers more flexible codec preference settings than earlier releases. Enterprise Manager converts your existing codec configuration to new 7.5 codec profile settings. Because of this, after you upgrade to release 7.5, you should review the converted codec settings and adjust them as needed.

G.722 Supported Devices/Interfaces

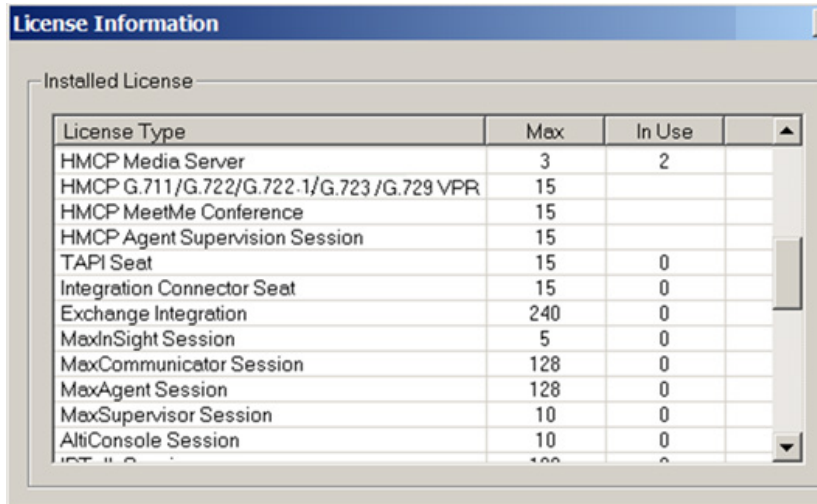
The G.722 codec is supported in HMCP media servers only. AltiGen's telephony boards, including the Triton and Vision boards, do not support the G.722 codec. G.722 is supported by the following devices:

- **AltiGen IP 805 phone** – G.722 64kbps ONLY
- **Polycom phones**
 - SoundPoint IP-335, IP-550, IP-560, IP-650, IP-670
 - SoundStation IP-6000, IP-7000
 - VVX 300, VVX 310, VVX 400, VVX 410, VVX500, VVX 600
- **SIP tie trunks**

NET and AudioCodes devices do NOT support G.722.

G.722 License Information

The G.722 codec is enabled using the Combo codec license. The number of total codecs is shown in the License Information panel (for Softswitch and Cloud deployments).



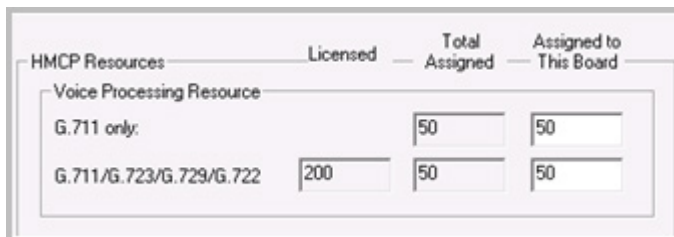
License Type	Max	In Use
HMCP Media Server	3	2
HMCP G.711/G.722/G.722.1/G.723/G.729 VPR	15	
HMCP MeetMe Conference	15	
HMCP Agent Supervision Session	15	
TAPI Seat	15	0
Integration Connector Seat	15	0
Exchange Integration	240	0
MaxInSight Session	5	0
MaxCommunicator Session	128	0
MaxAgent Session	128	0
MaxSupervisor Session	10	0
AltConsole Session	10	0

You will see the G.722 codec choice in various places throughout MaxAdministrator, wherever you see the other codec choices for Softswitch and Cloud deployments.

G.722 in HMCP Board Configuration

In Release 7.5, the G.722 codec has been added to Voice Processing Resource choices for HMCP Boards (for Softswitch and Cloud deployments).

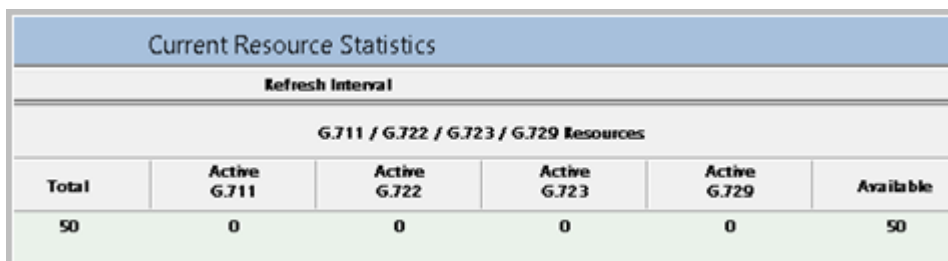
The combo codec resources allocated in the system are used to process this codec, in addition to G.723 and G.729.



HMCP Resources	Licensed	Total Assigned	Assigned to This Board
Voice Processing Resource			
G.711 only:		50	50
G.711/G.723/G.729/G.722	200	50	50

G.722 in the Current Resource Statistics View

In MAXCS 7.5, you will see G.722 in the Current Resource Statistics view (for Softswitch and Cloud deployments).

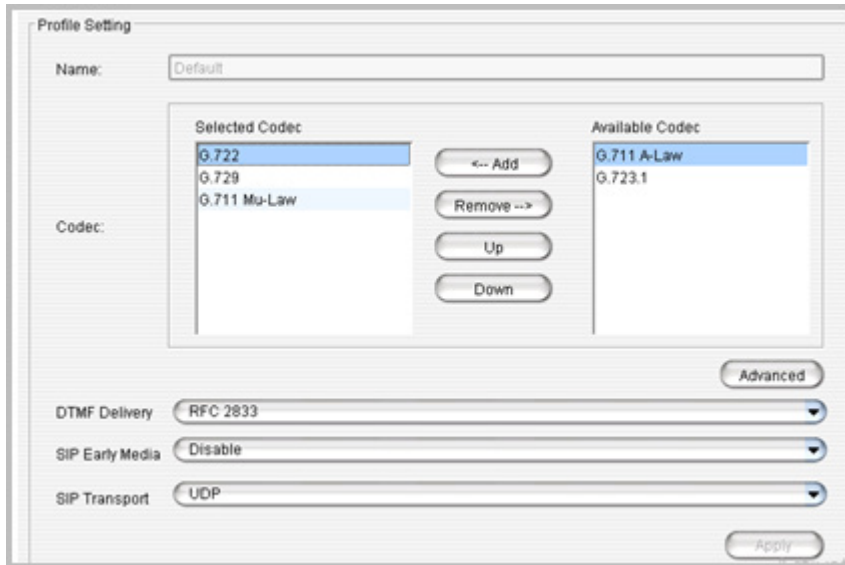


Current Resource Statistics					
Refresh Interval					
G.711 / G.722 / G.723 / G.729 Resources					
Total	Active G.711	Active G.722	Active G.723	Active G.729	Available
50	0	0	0	0	50

Managing Codec Priority

The Enterprise Manager *Codec* tab has been updated. You'll now see a list of available and selected codecs. Click the **Add** and **Remove** buttons to move codecs from one list to the other, as needed.

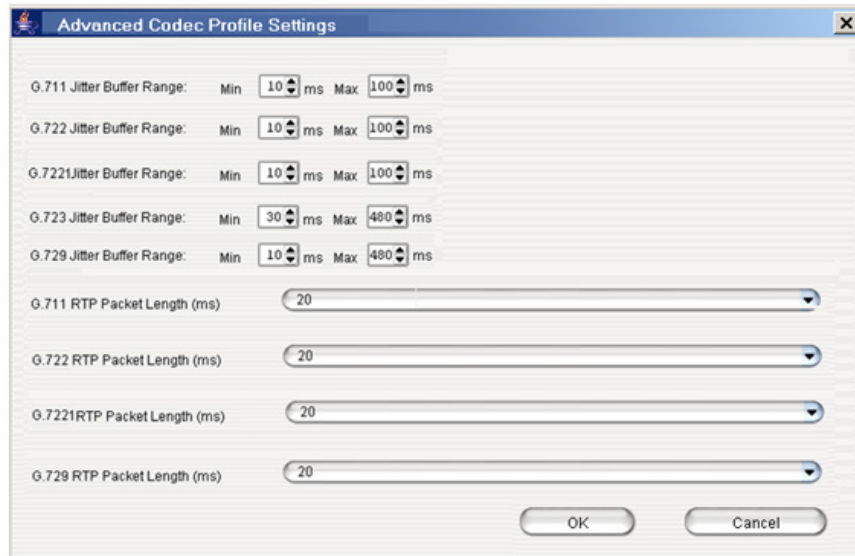
During codec negotiation, the highest-priority codec that the receiving end supports will be used. Therefore, you should sort the *Selected Codec* list by priority, from top to bottom. Use the **Up** and **Down** buttons to do this.



The *Selected Codec* list also appears in the SDP Codec Capability list for codec negotiation.

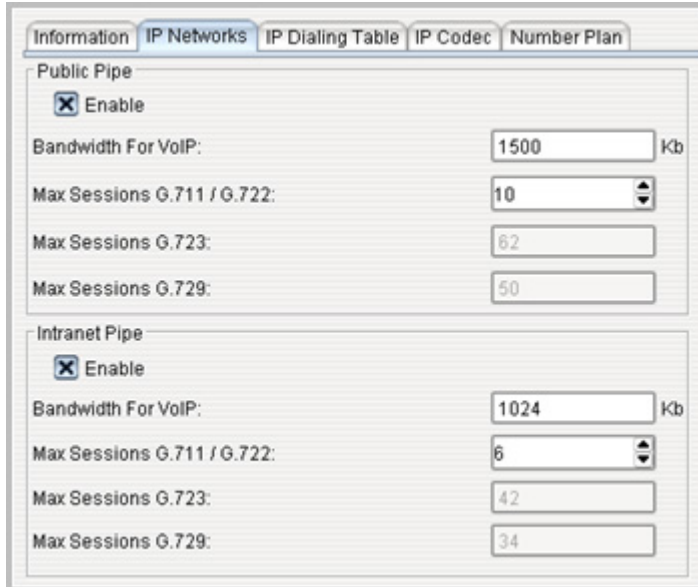
The codec profile assigned to the SIP-TIE Trunk must use RFC2833 (DTMF) in order to support Polycom phones. We recommend that you configure a single codec for SIP Trunk interfaces.

Click the **Advanced** button to set codec Jitter Buffer Range and codec RTP Packet Length (ms) for each codec.



Managing IP Network Bandwidth

Also in Enterprise Manager, the **Servers > IP Networks** tab has been updated.



Section	Enable	Bandwidth For VoIP (Kb)	Max Sessions G.711 / G.722	Max Sessions G.723	Max Sessions G.729
Public Pipe	<input checked="" type="checkbox"/>	1500	10	62	50
Intranet Pipe	<input checked="" type="checkbox"/>	1024	6	42	34

Instead of specifying the maximum number of sessions per each codec, as in previous releases, in release 7.5 you specify the maximum number of sessions per *bandwidth range*.

The actual network bandwidth required for the bit rate range is used for the bandwidth calculation. 33 to 64 Kbps bit rate is translated into 100Kbps network bandwidth, which includes LAN and IP/UDP/RTP header overhead. 16 to 32Kbps and under 16Kbps are translated into 65 Kbps and 40 Kbps, respectively.

The maximum number of allowed G.711 and G.722 codecs appears by default. You can lower these numbers by clicking the down arrow until you reach the desired amount.

After the maximum number of sessions has been reached, the call negotiated with this codec bit rate will be rejected. Also, after the total bandwidth utilization of all the codecs exceeds the value specified in Bandwidth for VOIP parameter, new calls cannot be established.

Codec Negotiation

The logic for negotiating codec capabilities varies for various types of connection.

Codec Negotiation – Server to Client

With a server-to-client connection, the highest priority codec in the server's codec list that is supported by the client will be assigned.

Following is an example of a typical setting for SIP devices on a LAN: G.722, then G.711 Mu-Law. This hierarchy allows Polycom phones and the AltiGen IP-805 phone to connect via G.722. AltiGen IP phones will connect with G.711.



The screenshot shows a configuration window with two columns: "Selected Codec" and "Available Codec". In the "Selected Codec" column, G.722 and G.711 Mu-Law are listed. In the "Available Codec" column, G.711 A-Law, G.723.1, and G.729 are listed. Between the columns are four buttons: "<-- Add", "Remove -->", "Up", and "Down".

The next example shows a typical setting for SIP devices on a WAN: First G.729, the G.723.1, and then G.711 u-Law. This hierarchy allows remote SIP devices to use the compressed codec G.729. Remote IPTalk instances use G.723.1, and G.711 is used for other remote devices that do not support the other two codecs.



The screenshot shows a configuration window with two columns: "Selected Codec" and "Available Codec". In the "Selected Codec" column, G.729, G.723.1, and G.711 Mu-Law are listed. In the "Available Codec" column, G.711 A-Law and G.722 are listed. Between the columns are four buttons: "<-- Add", "Remove -->", "Up", and "Down".

Codec Negotiation – Server to Server

With server-to-server connections, the responding device determines the codec from the common codecs that both of the devices accept. Usually, the responding device selects the first preferred codec from among the common list. We recommend that you assign a preferred codec on both servers, so that both incoming and outgoing events use the same codec.

If you are not sure of the codecs supported by your SIP Trunk provider, then we recommend that you configure a single codec (your preferred codec) for SIP Trunks.

Codec Upgrades

As mentioned earlier, Release 7.5 offers more flexible codec preference settings than earlier releases.

Enterprise Manager converts your existing codec configuration to new 7.5 codec profile settings. Because of this, after you upgrade to release 7.5, you should review the converted codec settings and adjust them as needed.

The following table details how codec profiles are converted during the upgrade process.

Original Codec Profile	Converted MAXCS 7.5 Selected Codec Profile
G.711 Mu-Law	G.711 Mu-Law
G.711 A-Law	G.711 A-Law
Prefer G723.1 support G729	G723.1, G729
Prefer G729 support G723.1	G729, G723.1
Prefer G.711 Mu-Law support G.711 A-Law	G.711 Mu-Law, G.711 A-Law
Prefer G.711 A-Law support G.711 Mu-Law	G.711 A-Law, G.711 Mu-Law

Codec Preference (HMCP)

MAXCS 7.5 includes a new feature designed to help you reduce the CPU consumption that occurs as a result of codec encoding and decoding. It allows you to indicate a codec preference (G.729 or G.711) for calls handled via a SIP trunk. This approach eliminates the steps of encoding and decoding – packets are directly forwarded to the endpoint.

The preference that you set must be supported by the SIP Trunk provider, and must be included in the codec profile list for SIP Trunks.

While this feature is enabled, then the SIP device's codec configuration in Enterprise Manager will be ignored.

Codec preference applies to HMCP only.

All SIP devices must support G.711 uLaw; if there are no common codecs on the device side, and then G.711 uLaw will be used.

This codec preference applies to all SIP Trunk inbound calls and direct extension to SIP trunk outbound calls that are initiated from a phone or dialed via a MAXCS client.

Fax-over IP overrides this setting; it will always use G.711.

Outbound calls initiated by the system (such as calls from the voicemail system or ONA) may not use the preferred codec.

To configure this feature,

1. Log into MAXCS Administrator with the superpassword. If you do not log in via the superpassword, the feature will not be enabled.
2. Open the HMCP board configuration panel (double-click **HMCP** in the **Boards** window).
3. To enable the feature, check the box and specify which codec to use (G.729 or G.711 Mu-Law). Click **Ok**.

Codec Preference

Enable codec preference for all SIP trunk inbound and direct outbound calls

Preferred Codec: G.729
G.711 Mu-Law

Codec Preference – Incoming Calls

When the Codec Preference feature is enabled, the system uses the following logic for handling incoming calls, based upon the content of the first SIP INVITE request.

Preferred Codec is G.729	
First SIP INVITE Request Content	Codec Used for Call
SIP Trunk supports G.729	The IP Codec table of Enterprise Manager is ignored. The system uses codec G.729 to negotiate with the endpoint.
SIP Trunk does not support G.729	No codec is enforced.
Preferred Codec is G.711 Mu-Law	
First SIP INVITE Request Content	Codec Used for Call
SIP Trunk supports G.711 Mu-Law	The IP Codec table of Enterprise Manager is ignored. The system uses codec G.711 Mu-Law to negotiate with the endpoint.
SIP Trunk does not support G.711 Mu-Law	No codec is enforced.

If the target extension or DNIS number is enabled for Fax-Over-IP (FoIP), then no codec is enforced.

Codec Preference – Outgoing Calls, Third-Party IP Phones

When the Codec Preference feature is enabled, the system uses the following logic for handling outgoing SIP trunk calls from third-party IP phones. (Calls are considered SIP trunk calls if the target has a SIP trunk access code or an outcall routing access code prefix.)

Preferred Codec is G.729	
First SIP INVITE Request	Codec Used for Call
Supports G.729	The IP Codec table of Enterprise Manager is ignored. The system uses codec G.729 to negotiate with the endpoint.
Does not support G.729	No codec is enforced.

Preferred Codec is G.711 Mu-Law	
First SIP INVITE Request	Codec Used for Call
Supports G.711	The IP Codec table of Enterprise Manager is ignored. The system uses codec G.711 to negotiate with the endpoint.
Does not support G.711	No codec is enforced.

If the source extension is enabled for Fax-Over-IP (FoIP), then no codec is enforced.

Codec Preference – Outgoing Calls, AltiGen IP Phones

Because AltiGen IP Phone SIP call requests are always from MAXCS, the behavior is different from calls from third-party phone extensions.

When a user makes an outgoing call from an AltiGen IP phone, the system follows the IP Codec table in Enterprise Manager. All AltiGen IP phones support G.729 and G.711 Mu-Law.

When the Codec Preference feature is enabled, the system uses the following logic for handling outgoing SIP trunk calls.

Preferred Codec	Codec Used for Call
G.729	The system modifies the IP phone's codec in the RE-INVITE SDP body with G.729 codec. After the modification, MaxCS must pick G.729 and G.711 codecs as the preferred codecs to negotiate with the endpoint.
G.711 Mu-Law	The system modifies the IP phone's codec in the RE-INVITE SDP body with G.711 codec. After the modification, MaxCS must pick G.711 and G.729 codecs as the preferred codecs to negotiate with the endpoint.

If the source extension is enabled for Fax-Over-IP (FoIP), then no codec is enforced.

Media Enhancements

The following media enhancements have been added for MAXCS 7.5:

- Media Pass-through
- T.38 Pass-through
- Fax Support
- Fax Routing

Media Pass-through (HMCP)

Another enhancement for system performance and voice quality is the Media Pass-through feature. This feature applies only to HMCP Softswitch systems; it is enabled by default.

While the Media Pass-through feature is enabled, the HMCP driver doesn't need to do encoding and decoding on both channels. This approach has the following benefits:

- Improved system performance, because no MIPS is required for codec processing
- Improved voice quality, because no distortion is introduced by additional compressed codecs
- Reduced voice latency, because it eliminates the delay introduced by codec and jitter buffer processing

In addition to benefits to direct calls, pass-through applies during call recording, silent monitoring, and coaching.

Pass-through is enabled by default. To disable it, modify the <PacketForward> statement in this file:

```
..\Altiserv\SP\HMP\HMPSP.xml
```

The system will reset the HMCP board so that the setting can take effect.

Notes:

- RTP headers are reconstructed before forwarding the packets to the target channel
- Allow SRTP enabled on one leg and disabled on another
- Require encryption/decryption processing if SRTP is enabled
- No Media pass-through support for playing and conference media streams

Conditions Not Supported

Media pass-through cannot support all conditions in the HMCP system, even if both connected channels are using same codec. The following MAXCS features do not supported media pass-through:

- Call playing: The trunk call playing must use encoder and decoder on RTP channel
- Conference calls – All RTP channels in a conference bridge must be encoded and decoded voice

T.38 Pass-through

T.38 is a standard for real-time fax transmission over IP networks. T.38 Pass-through relays two T.38 sessions between the SIP trunk and MAXCS and MAXCS gateway.

- T.38 pass-through is not supported on hardware chassis deployments
- T.38 pass-through is supported only on systems using AltiGen-provided SIP trunks
- T.38 pass-through is supported on these ATA devices: AudioCodes MP-202, MP-11x, and MP-124

See the next two sections for enabling FoIP for extensions and In Call routing.

Fax-over-IP (FoIP) Extensions

Release 7.5 supports Fax-over-IP (FoIP), leveraging T-38 pass-through. Both voice calls and fax calls can be made through a FoIP extension; both incoming and outgoing calls on the extension will use G.711 codec.

Fax extension-to-extension is supported; the SIP endpoints must be a supported ATA device.

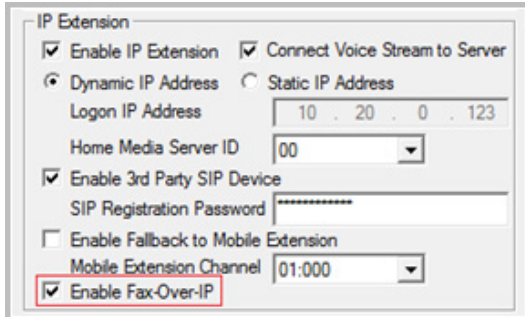
Limited FoIP extension parameters are supported:

- First/Last Name, Password, Description, Department
- DID Number, Transmitted Caller ID, E911 CID
- IP Extension parameters, except Enable Fallback to Mobile Extension
- Out Call Restrictions

- Forward All Calls – forwarding target must be an FoIP extension or FoIP hunt group
- Busy Call Handling – forwarding target must be an FoIP extension or FoIP hunt group

To configure an extension as a FoIP extension,

1. In MAXCS Administrator, select **PBX > Extension Configuration**.
2. Select the extension on the left.
3. On the right, in the *IP Extension* group, check **Enable Fax-Over-IP**.



Once Fax-Over-IP has been enabled, all outbound calls from this IP extension will use G.711 codec, and all incoming DID routing calls to this extension will use G.711 codec.

FoIP In Call Routing

Release 7.5 supports support two different In Call routing methods for Fax numbers:

- DNIS In Call Routing
- FoIP Extension DID number

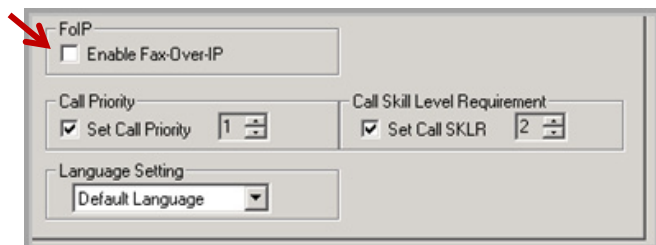
All FoIP inbound and outbound voice calls use G.711; all other codec settings are overridden.

Routing supports T.38 UDPTL (UDP packets) using the same RTP port range.

TKPT (TCP based) is not supported.

To enable FoIP DNIS In Call routing,

1. In MAXCS Administrator, select **PBX > In Call Routing Configuration**.
2. Select the number.
3. On the DNIS Routing tab, check **Enable Fax-Over-IP** and click **OK**.



Once this option is enabled, then, all incoming calls which match this entry will use G.711 codec. If this option is not enabled, but the destination is a fax Extension, G.711 codec will be used.

FoIP Hunt Group

MAXCS 7.5 supports FoIP Hunt groups; there is no specific configuration type for such hunt groups. In a FoIP hunt group, all group members must be FoIP extensions. Group queuing is not supported.

Limited FoIP group parameters are supported:

- First/Last Name, Password, Description, Department
- DID Number
- Ring First Available Member and Ring Next Available Member
- Busy Call Handling and No Answer Handling should be disabled
- Forwarding All Calls – forwarding target must be a FoIP extension or FoIP Hunt group

Configure FoIP Hunt groups as you would any other type of Hunt group; include only FoIP extensions.

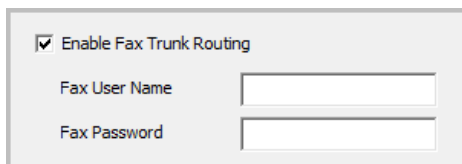
Fax Routing

To simplify administrative tasks, you can allow voice and FAX calls to run on the same SIP trunk channel. The SIP trunks must be configured to support both voice and FAX.

The SIP trunk uses the same SIP server IP address, but different authentication credentials for voice trunk versus a fax trunk.

To configure fax routing,

1. In MAXCS Administrator, open Boards view and double-click **SIPSP**.
2. Click **Board Configuration**, then **SIP Trunk Configuration**, then **SIP Trunk Profile**.
3. For the voice SIP Trunk, check **Enable Fax Trunk Routing** and enter the fax trunk's user name and password. Click **OK**.



The screenshot shows a configuration window with a checked checkbox labeled "Enable Fax Trunk Routing". Below it are two input fields: "Fax User Name" and "Fax Password".

Configure Out Call routing just as voice and FAX are supported in the same SIP trunk.

For AltiGen SIP trunks, you must configure one SIP trunk channel to perform SIP registration for GW1 and GW2 of voice trunk and GW1 and GW2 of the fax trunk individually. Refer to the *Migration, SIP Trunk Configuration, and Testing Guide* for Release 7.5 for configuration details (when available).

SNMP Enhancements

MAXCS 7.5 includes many additional traps to enhance event monitoring:

- SIP Trunk Group Status traps (Trunk Group Up, Trunk Group Down, Trunk Group Busy)
- SIP Abnormality traps (Out of HMCP codec resources, insufficient SIP extension channels, malicious IP address added)
- Security traps (Toll fraud – outbound call from voicemail exceeding call duration threshold or call quality threshold, 911 call, extension lockout)
- Out of licenses traps

SNMP traps are logged in text files: SNMP_Altiserv.txt, SNMP_DinaProxy.txt, SNMP_GlobalAppearance.txt, and SNMP_SPServ.txt.

Additional Enhancements

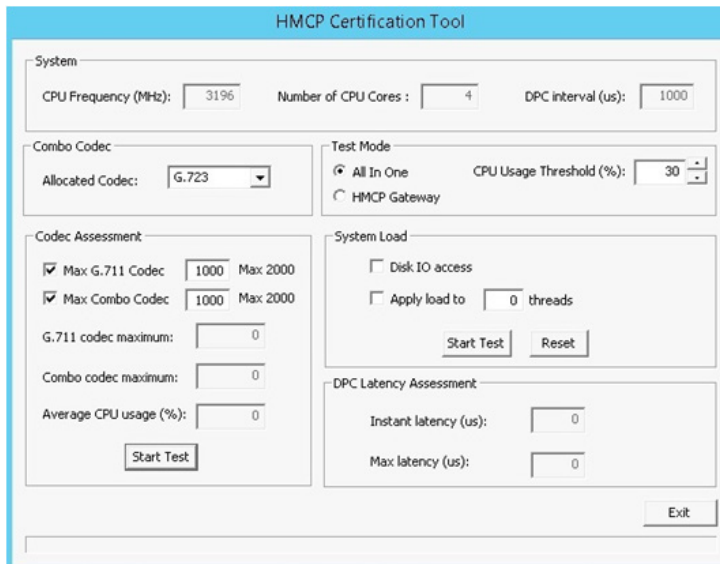
MAXCS 7.5 includes a few additional enhancements.

- The HMCP Certification tool has been expanded
- The Block Unauthorized SIP Invite feature now has a checkbox within MAXCS Administrator
- A Version number field has been added for tighter SightMax integration.

HMCP Certification Tool Enhancements

MAXCS 7.5 includes several improvements to the HMCP Certification tool:

- Support for G.722 codec
- DPC latency measurement (instant latency and maximum latency)
- System load (Disk I/O access, apply load to thread)



The screenshot shows the HMCP Certification Tool interface with the following sections:

- System:** CPU Frequency (MHz): 3196, Number of CPU Cores: 4, DPC interval (us): 1000
- Combo Codec:** Allocated Codec: G.723
- Test Mode:** All In One, CPU Usage Threshold (%): 30, HMCP Gateway
- Codec Assessment:** Max G.711 Codec (1000 Max 2000), Max Combo Codec (1000 Max 2000), G.711 codec maximum: 0, Combo codec maximum: 0, Average CPU usage (%): 0, Start Test button
- System Load:** Disk IO access, Apply load to 0 threads, Start Test, Reset buttons
- DPC Latency Assessment:** Instant latency (us): 0, Max latency (us): 0
- Exit:** Exit button

Block Unauthorized SIP Invite

Release 7.5 expands the registry entry that was added in Release 7.0 Update 1. You now can block unauthorized SIP invite messages by selecting a checkbox. This setting is disabled by default.

When you enable this setting, the SIP Invite requests are ignored if IP address is not configured in one of the following places:

- SIP trunk
- IP dialing Table
- Trusted SIP Device List

IP extensions for the following devices are added to the Trusted SIP Device list automatically:

- AltiGen IP Phone
- 3rd Party SIP Device
- IPTalk

To enable protection,

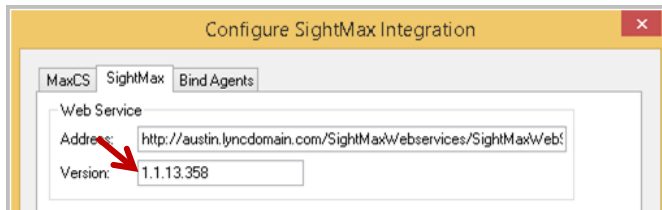
1. In MAXCS Administrator, open Boards view and double-click **SIPSP**.
2. Select **Board Configuration > Advanced Configuration**.
3. Check **Block Unauthorized SIP Invite** and click **OK**.



SightMax Version Setting

MaxCS 7.5 includes a new field for the SightMax Integration application. The version number that is entered will be checked when the system connects to a SightMax server.

This field is located on the SightMax tab of the integration application.



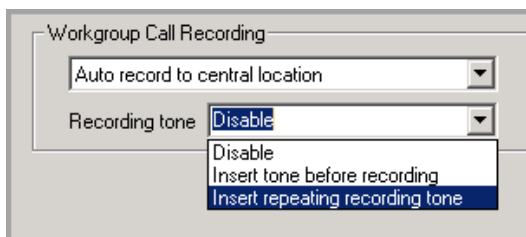
Repeating Recording Tone

A new field in release 7.5 allows workgroups to set a repeating recording tone. When this is set, workgroup calls beep once every 15 seconds.

Note: This feature works only when one or both endpoints are SIP; for example, from a PRI trunk to a SIP extension.

To configure this repeating tone,

1. In MAXCS Administrator, select **Call Center > Workgroup Configuration**.
2. Select the workgroup.
3. On the *General* tab, set *Recording Tone* to **Insert Repeating Recording Tone** and click **OK**.



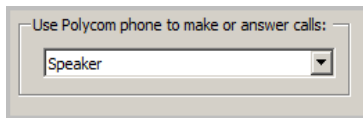
Client Enhancements

MaxAgent and MaxCommunicator have new settings, to better support Polycom phones.

Handling Calls via Clients (Advanced License required, VVX only)

Users can configure MaxAgent and MaxCommunicator to answer or place calls via your Polycom phone.

In MaxAgent and MaxCommunicator, the setting is under **Configuration > Extension: Use Polycom phone to make or answer calls**.



There are three options for this feature:

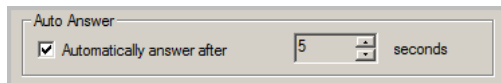
- **Disabled** – MAXCS rings back the Polycom phone, and you must manually answer the call.
- **Speaker** – When you click an incoming call from the MAXCS client, the call will be answered by the phone's speaker; outgoing MAXCS client calls will be dialed and directed to the phones' speaker.
- **Headset** – When you click an incoming call from the MAXCS client, the call will be answered by the phone headset; outgoing MAXCS client calls will be dialed and directed to the phones' headset.

This answering feature is enabled only when your phone is on-hook.

Auto-Answer via Clients (Advanced License required, VVX only)

This Auto-Answer feature is available only when the **Use Polycom phone to make or answer calls** option is set to **Speaker** or **Headset**.

In MaxAgent and in MaxCommunicator, the Auto-answer setting is under **Configuration > Extension > Call Handling**.



Users can check the **Automatically answer after xx seconds** checkbox, and set the number of seconds before calls are picked up. Note that there may be a few seconds of delay answering calls with this feature.

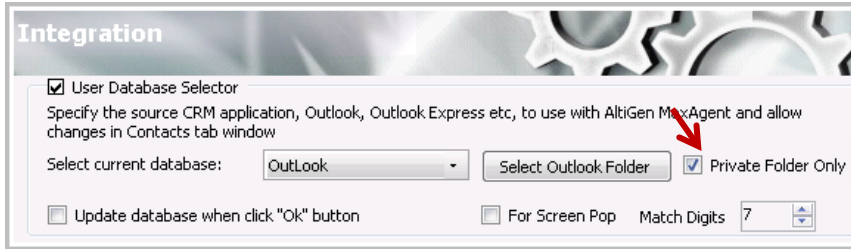
Synchronize the Private Contacts Folder Only

If an organization has many large public contact folders for Outlook, users can eliminate the time it takes to synchronize all of those folders by checking a new option.

In MaxAgent and MaxCommunicator, users can control whether to synch both public contact folders and private contact folders.

Within MaxAgent,

1. Click the **Configuration** button and select **CRM Integration**.
2. Select **Outlook** for the current database.
3. Check the **Private Folder Only** checkbox to synchronize only the Private Contact folder. Clear the checkbox to also synchronize Public Contact folders.



AltiGen Technical Support

AltiGen does not provide general configuration support for AudioCodes products. For general configuration information, refer to your AudioCodes documentation.

AltiGen provides technical support to Authorized AltiGen Partners and distributors only. End user customers, please contact your Authorized AltiGen Partner for technical support.

Authorized AltiGen Partners and distributors may contact AltiGen technical support by the following methods:

- You may request technical support on AltiGen's Partner web site, at <https://partner.altigen.com>. Open a case on this site; a Technical Support representative will respond within one business day.
- Call 888-ALTIGEN, option 5, or 408-597-9000, option 5, and follow the prompts. Your call will be answered by one of AltiGen's Technical Support Representatives or routed to the Technical Support Message Center if no one is available to answer your call.

Technical support hours are 5:00 a.m. to 5:00 p.m., PT, Monday through Friday, except holidays.

If all representatives are busy, your call will be returned in the order it was received, within four hours under normal circumstances. Outside AltiGen business hours, only urgent calls will be returned on the same day (within one hour). Non-urgent calls will be returned on the next business day.

Please be ready to supply the following information:

- Partner ID
- AltiGen Certified Engineer ID
- Product serial number
- AltiWare or MAXCS version number
- Number and types of boards in the system
- Server model
- The telephone number where you can be reached