



Release Notes: SMR.2.162.234



@COM Business Manager v2

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This release adds new features, changes, and bugs fixes for the @COM Business Manager v2. This SMR release can be installed on all systems running at least SMR.2.1.185 and therefor also includes all previous updates since:

- SMR.2.87.127 ([Online change log](#))
- SMR.2.146.201 Beta ([Online change log](#)) **NOTE: This update was never publicly available**
- SMR.2.162.234 ([Online change log](#))

The update will determine automatically whether it needs to install any of the older updates too.

1. New Features

Shown in the next paragraphs is a more elaborate explanation of new features in this SMR release; however, some smaller features might not be listed here. For a full list of all new features, changes and resolved bugs please check the @COM website at <https://issues.technetworks.eu/>.

1.1 Fixed-Mobile extensions

The @COM now offers so called "fixed-mobile extensions" which are extensions that forward calls to extension fixed or mobile phones, allowing external phones to act as internal extensions. These extensions can be compared to "virtual extensions with call-forwarding" with the exception that these extensions can also be assigned to a user and virtual extensions cannot. Being able to assign the "fixed-mobile extension" to a user, enables this user to use the @COM CTI Client and/or @COM Personal Call Manager in combination with their fixed- or mobile phone. In future we also plan to also support fixed-mobile extensions to be used as Call Center Agents, but as for now this is not yet possible.

Fixed-Mobile extensions can also be used in combination with the "Forced-on-PBX" feature that is used by certain providers, like oneCentral AIR, to route all calls from and to a mobile phone via our PBX. This enables an even further integration of the mobile phone into your telephone system.

IMPORTANT: Changed required for existing "Forced-on-PBX" (a.k.a. "Fixed-Mobile"/"VAMO") Users! Due to the lack of the "Fixed-mobile" feature in earlier releases of the @COM Business Manager v2 we provided customers with a free license for the "Forced-on-PBX custom-script" enabling them to use the Forced-on-PBX solution whilst we did not yet provide full-support for it. We instructed all resellers to nevertheless sell their customers the fixed-mobile device license as we would enable these licenses once the feature became available and then revoke the free license for the "Forced-on-PBX custom-script". With the arrival of the "fixed-mobile extension" in this release that time has now come. Which means:

All users that now use the free "Forced-on-PBX custom-script" must be converted to fixed-mobile extensions before this free license is revoked.

We will allow resellers and customers 6 months' time from the day of this release to convert all their "Forced-on-PBX" users, after which all free "Forced-on-PBX custom-script" license will be revoked. Revoking can also occur after a reseller has indicated to us that it has converted all their users.



1.2 New Personal Call Manager

We have completely revised the @COM Personal Call Manager in such a manner that it now looks more like an App and used HTML5 instead of HTML4. This change allows us to better integrate with web-browsers and create a Mobile App version of the PCM, which is now also available (for free). We added some extra features to the PCM as well:



- Use a fixed-mobile device (besides only a normal or flex devices) as the extension for the PCM.
- Send your office Caller ID when using your private fixed-mobile device.
- Listen and manage voicemail messages and greetings.
- View Real-time status of @COM users and extensions.
- Direct dial of phone numbers via the built-in keypad. (*)
- Answer, hang-up, place on hold and transfer calls using the PCM. (*)
- Choose the device you want to use for incoming and outgoing calls. (*)
- Dial a number in nearly any format. (No zero for an outside line required). (*)

For more info: <https://docs.at-com.nl/en/bm/apps/pcmv2>

(*) This requires the new "Call Control" license/feature.

The "Call Control" license/feature already existed in the @COM Business Manager v1.3 but has long been replaced by the "@COM CTI Level" licenses. Within the @COM Business Manager v2 the "Call Control" has returned but with a whole different purpose.

1.3 New Personal Call Manager App

The @COM Personal Call Manager App (aka: PCM App) is a FREE add-on for the web-based @COM Personal Call Manager v2 (aka: PCM) with all the advantages of your mobile phone or tablet. The @COM Personal Call Manager v2 can be used on a mobile phone via its web-browser, but the App offers a further integration with your mobile phone offering a much better fixed-mobile solution with unique properties. All features present in the @COM Personal Call Manager v2 are off course also present in the PCM App, but are enriched with the following features:



- Access your personal contacts on your mobile phone or tablet from within the PCM App. (*)
- Transmit to the @COM Business Manager whether you are in a call or not, allowing your colleagues to see if you are on the phone or not. (*)
- Choose if you want to dial-out via the @COM Business Manager or your own mobile network.

For more info: <https://docs.at-com.nl/en/bm/apps/pcm/app>

IMPORTANT:

To be able to use the PCM App you **MUST** connect to the @COM Business Manager over a secure HTTPS connection with a valid SSL certificate or else the PCM App will not work. Self-Signed certificates will not work properly as they are blocked by the operating system of the mobile phone or tablet.

(*) Due to privacy these features can be turned off and on by the user without notification.

The PCM App only communicates the phone number you which to dial and if enabled the busy state of your mobile phone to the @COM Business Manager. Whom you are calling with, or even if the phone call as taken place is not logged anywhere when you are not calling via the @COM Business Manager. For more details regarding your privacy, please check our [Privacy Policy](#).



1.4 User based call routing

We have introduced a new way of routing calls to the correct destination, normally we would assign a DID to an extension and assigned to that extension is a device. This hasn't not changed, but once an extension has been assigned to a user (as the main extension), we route the call to the extension defined by the user. The user can now select on which extension he/she wants to be available using the @COM Personal Call Manager (PCM) and an @COM Administrator decides from which extensions the user can choose.

In the CDL lists you can now also select the destination; "User". This enables you to force the routing based on the choice of that user in the PCM. However, the common practice will be to route the call to the main extension of the user which can also be achieved by selecting "*Originally Called Extension (*)*".

NOTE:

After updating to this SMR version, all existing users will only have the originally assigned extension as their main extension. Additional extensions will have to be added manually by an @COM Administrator. The same applies to users automatically created via an Active Directory synchronization.

1.5 Full Support for newer Snom and Mitel phones

Due to the change described in chapter 2.1 we can now offer Full-Support for even more Snom and Mitel phone models. In total we added support for 1x Aastra, 1x Mitel and 11x Snom phones including support for the latest Snom v10 firmware and Mitel v5 firmware.

All the phone models which we now fully support are listed here;

Aastra

- 6730i, 6731i, 6735i, 6737i, 6739i*, 6753i, 6755i and 6757i.
- *with key modules:* M670i and M675i

Mitel

- 6863i, 6865i, 6867i, 6879i and 6873i*
- *with key modules:* M680i and M685i

OpenStage

- 15, 20, 20E, 40, 60 and 80 (with specific firmware and not behind a NAT)
- *with key modules:* SLK1 and SLK2

Snom

- 300, 320, 360 and 370
- 710, 715, 720 and 760
- D120*
- D305*, D315*, D335*, D345*, D375 and D385*
- D712*, D715, D717*, D725, D735*, D745*, D765 and D785*
- *with key modules:* SXM, UXM-D3, UXM-D7 and UXM-D7C

(*) new models



All existing devices that were created in the past using the wrong type will be automatically updated to the correct device type during the update. However, these devices will receive a default key programming because programming of function keys on the unsupported devices was not possible.

IMPORTANT:

After updating to this new release make sure to manually update the key-programming of all previously unsupported devices. The update will not alter the currently programmed keys on these devices, but if you change settings on these devices some day later using the LMT, it might cause all function keys to be reprogrammed to what would be an empty template if you didn't perform this action.

1.6 Audio and Display Language can be set per device

In the past there was a single language setting for specific device types which would only change the display language, or audio language, or both. Now it's possible to set the audio language on all device types and set the display language on all fully supported devices.

1.7 Online LMT documentation from within the LMT

From within anywhere in the LMT you can now perform a mouse-over on fieldname of any setting. The mouse-over will cause a small pop-up that displays information about the specific setting directly from the new website: <https://docs.at-com.nl>. In the pop-up there is also a link that redirects you to the full page with all other settings at: <https://docs.at-com.nl> which is now the online manual for the LMT and much more to come.

NOTE:

During the release of this firmware most pages on <https://docs.at-com.nl> have been written but have not yet been published by the chief-editor. Please allow our chief-editor some time to review and publish all the new pages.

We expect to have most pages online by the end of Q1 2022.

1.8 New ACL's for PCM features

Because of the new PCM we expect to see an increase of public IP traffic destined for the PCM webpages. To better control the access of this HTTPS service we have added new Access Control Lists (ACL). The new ACLs are named:

- Address book - Get phonebook contacts
- Call Control - Check Login and Make Call
- Extension - Enable/Disable DND and Call Forwarding
- Voicemail - View/edit Configuration, messages, greetings and move/delete messages

If you plan on using the new PCM over public networks make sure to change these access-lists from "10.0.0.0/8, 172.16.0.0/12, 192.168.0.0/16" to "0.0.0.0/0".

NOTE: These names will change in the future, but for now these are their names.



1.9 New device option: "Allow DTMF transfers"

All devices now have an additional setting on the "SIP" tab called "Allow DTMF transfers". When this setting is enabled, you can perform a call transfer using the DTMF tones:

- ***0** Disconnect the current call and return to waiting caller if available.
- ***1** Initiate a blind transfer.
- ***2** Initiate an announced transfer.
- ***3** Swap between the active call and a waiting caller.
- ***4** Initiate a conference between the active call and the waiting caller. (3-way conference)
- ***5** Complete an announced transfer (same as just hanging up during transfer)

This feature is a solution to circumvent the licensing limitation on certain free soft-clients, but also to circumvent the issues with Microsoft Teams client (Voice 365) that will remain busy after a call-transfer because Microsoft transfers the call externally and not internally. But it is also required to perform announced call-transfers using the PCM on devices that are not fully supported or behind a NAT.

IMPORTANT:

This feature is required on all devices that are NOT fully supported to allow announced call transfer by the PCM. If not enabled the announced call transfer option will be disabled in the PCM. The feature should also be enabled on fully supported devices which are located behind a NAT!

But be aware that enabling the option will disable route optimisation!



2. Changed functionality

Shown in the next paragraphs is a more elaborate explanation of changed functionality in this SMR release; however, some smaller changes might not be listed here. For a full list of all new features, changes and resolved bugs please check the @COM website at <https://issues.technetworks.eu/>.

2.1 New function key programming for Fully Supported devices

We have completely rewritten the GUI part of the device management in the LMT for fully supported devices. This change was needed to make sure we could more easily add support for new phone models. Whilst building this new device management we also drastically altered the way you can program function keys on fully supported devices. This now allows a much more visual interface to program keys: Besides being more visual you can now also click on the keys in the image to program that specific key,

Programmable function keys. (fkeys)

Function Contextual Predefined

Function keys beside display:

Page: 1 2

Type	Value	Label
Voicemail Dir.		Pers. Telb
Corporate Dir.		Cent. Telb
VM listen		
DND		
Speed-dial	*51767	RP In/Uit
Speed-dial	*25	> GSM

Type	Value	Label
BLF	765	Pieter
BLF	763	Prewesh
Speed-dial	761	Nick (761)
Speed-dial	762	Rob (762)
-		
More		More

or select and update it on the right-hand side of the window. In addition to just being able to program the standard function keys, we now allow you to program every key the phone allows us to program which included the predefined keys and contextual keys during idle state (keys below the display). And probably one of the most appreciated features; you can now copy keys from one device to many other similar device types just like you were able to copy the device settings.



2.2 The Personal Call Manager GUI has been updated

We have launched a new Personal Call Manager as we already described in chapter 1.2. We mentioned it as a new feature as so much has been changed that it literally is a new Personal Call Manager. But it's actually a change, which is why we mentioned it here as well.

IMPORTANT: Advise existing PCM users of this [major change](#) before updating!
The new interface may be straight forward and self-explaining, but it's good to inform customers in advance or else they may get confused. Also, the old PCM cannot be accessed anymore after the update.

2.3 Improved Forced-on-PBX (VAMO v2) support

We have finally introduced the "fixed-mobile" extensions into the @COM Business Manager v2 which was originally planned for the first release. With this new feature (that is already described in chapter 1.2) we now no longer need the "Forced-on-PBX custom-script" originally designed for v1.3 systems and updated to work with v2 systems, and it allows us to add more control over the Caller ID being displayed on the mobile and internal extensions when exchanging calls which means you can now:

- Show the internal or external Caller ID on calls from a Forced-on-PBX device (VAMO) to an internal extension.
- Show the internal or external Caller ID of internal callers on calls to a Forced-on-PBX device (but only in case the SIP provider support this!).
- Send or not send the original Caller ID of an external caller to the Forced-in-PBX device.

Additionally, we have introduced an easier method to ensure that calls to Forced-On-PBX devices are routed via the correct route. In case "Caller ID routing" was enabled this limited the used of which Caller ID could be sent, as the selected Caller ID always had to be the public Caller ID.

IMPORTANT: Changed required for existing "Forced-on-PBX" (a.k.a. "Fixed-Mobile"/"VAMO") Users!
Due to the lack of the "Fixed-mobile" feature in earlier releases of the @COM Business Manager v2 we provided customers with a free license for the "Forced-on-PBX custom-script" enabling them to use the Forced-on-PBX solution whilst we did not yet provide full-support for it. We instructed all resellers to nevertheless sell their customers the fixed-mobile device license as we would enable these licenses once the feature became available and then revoke the free license for the "Forced-on-PBX custom-script". With the arrival of the "fixed-mobile extension" in this release that time has now come. Which means:

All users that now use the free "Forced-on-PBX custom-script" must be converted to fixed-mobile extensions before this free license is revoked.

We will allow resellers and customers 6 months' time from the day of this release to convert all their "Forced-on-PBX" users, after which all free "Forced-on-PBX custom-script" license will be revoked. Revoking can also occur after a reseller has indicated to us that it has converted all their users.

2.4 Call Transfer detection rewritten

The source code for detection of call transfers has been rewritten to allow support for mobile extensions. As a result, this may result into changed behavior into displaying the Caller ID on calls that were transferred using the "Mobilyt extender (DTMF transfer)" available on *25 forwarded calls.



2.5 The @COM Connect can now start without a license

To provide the new @COM Personal Call Manager with real-time data the @COM Connects is being used. Normally this would require an additional license to be installed but we have updated the @COM Connect software to allow it to operate without a license for use only with the PCM software.

The @COM Connect running without a license will however not accept any CSTA/XML III connections and can thus not be used for other purposes other than the PCM.

2.6 The Debug-mode has been optimized for larger systems.

The telephony server debug-mode could not be safely enabled on large systems with high loads because the debug-mode itself quadrupled the load on the system. We have now removed the debugging that was causing this high-load from the "debug-mode" because the output it produced was hardly ever used.

This now makes the debug-mode easier to apply on larger systems.

But keep in mind that the load is still increased significantly! (A factor 2x is to be expected)

2.7 A Standard administrator can now also create multiple devices

We have updated the access-rights allowing standard administrators to also be able to create multiple devices within the LMT (instead of only a single device, one at a time).

2.8 Improved support for detection of Semi-Blind transfers

Semi blind transfers are transfers that are initiated as an announced transfer, but whilst the call to the destination extension is still ringing, the user elects to transfer the call.

This transfer method is not supported by the @COM Business Manager!

We have been able to block the method on many devices, but on some devices it's not possible to block this method of transfer, especially on most soft clients. The problems it causes are not immediately visible to the user, but they cause havoc in the telephony server. This results in issues with CDR records, and consequently the CDR reports. But it also causes issues if the call is transferred a second time after being semi-blind transferred.



3. Bug fixes

Shown in the next paragraphs is a more elaborate explanation of major bug fixes in this SMR release; however, some smaller fixes might not be listed here. For a full list of all new features, changes and resolved bugs please check the @COM website at <https://issues.technetworks.eu/>.

3.1 Improved performance of the Asterisk Event Daemon

We noticed that the Asterisk Event Daemon was causing high CPU and database loads. On further inspection we found the cause for this and were able to bring down the loads significantly! As a result, larger systems now respond quicker especially when they have lots of extensions and/or call center queues, which is noticeable in applications like the Call Center Wallboard, Call Center Supervisor, Call Center Agent, PCM and Presence Board.

For more info see issue: <https://issues.technetworks.eu/view.php?id=5107>.

3.2 Could not change device license from "full-support" to "basic support"

This issue had been fixed.

For more info see issues: <https://issues.technetworks.eu/view.php?id=5005> and <https://issues.technetworks.eu/view.php?id=5216>

3.3 Databases were sometimes unreachable for Asterisk

The ODBC database drivers used by the telephony server (Asterisk) had some known issues and required updating. This has now been done and additionally we also enable multiple database connections (connection pooling) for the telephony server (Asterisk) so that instead of a single connection it can now use up to 10 connections in separate threads. This increases performance on systems handling multiple simultaneous calls and reduces the risk of SIP Deadlocks occurring due to smaller delays.

For more info see issue: <https://issues.technetworks.eu/view.php?id=4935>.

3.4 @COM Auto-update feature was not working

Updating @COM Business Manager v2 using scheduled or automatic updating was not working. This issue has now been fixed.

For more info see issue: <https://issues.technetworks.eu/view.php?id=4944>.

3.5 Incorrect detection of SIP Deadlocks could occur

Incorrect detection of SIP Deadlock may occur in these following scenarios:

- The SIP trunk is not included in the test because its name starts with numbers.
- If no peers are online.
- If three or less peers are online, only two or just one peer would be tested.
- If auto rotation of the logfiles coincides with the test.

3.6 Minor bug fixes

A small list of some minor bugs which we fixed and are still worth mentioning:

- The G.722 codec was not available for SIP trunks.
- Announcements were sometimes played at midnight as well.
- Call Center queues programmed as "Cyclic (linear)" were really set to "Cyclic from last agent".
- Could not view Device display when LMT was connected over HTTPS.
- The Voice recording archive feature could cause out of disk space scenarios.
- System Monitor e-mails did not honor the "Mail from (domain)" setting.
- Various CDR issues resolved.
- Various CLI errors resolved



4. Known issues

If you have access, please check out our issue-tracker at <https://issues.technetworks.eu/> for more info. Only this bug tracker will contain all known issues, only the major issues will be shown:

4.1 SIP Deadlocks causing auto-reboot of the telephony server

The cause for this issue has sadly not been identified let alone being able to reproduce. We very much hope that the new debug-mode will improve our chances of retrieving proper debug-traces from specific customers systems where this issue is occurring more often. But the new ODBC database connection for Asterisk as described in chapter 3.3 is expected to reduce the chance of SIP deadlocks occurring, however we are aware that this will not be the final solution.

For more info see issue: <https://issues.technetworks.eu/view.php?id=4753>

4.2 Sometimes DTMF tones are not recognized.

This issue occurs when using the RTP keep-alive feature. The work-around is to not use this feature and work-around the reason why you would need to use RTP keep-alive message in the first place.

For more info see issue: <https://issues.technetworks.eu/view.php?id=2401>

4.3 The COLP is not updated when retrieving a call from a parking position.

When a call is parked the Connected Line information (COLP) is not updated on the display of the caller when the call is retrieved from its parking position.

For more info see issue: <https://issues.technetworks.eu/view.php?id=2252>

4.4 Login time may not be visible in ACD Agent reports.

The Login time is not visible in an ACD Agent report when the agent was logged on during the full period selected in the report. This is because the report currently needs to "see" at least one login or logoff event within the selected period.

For more info see issue: <https://issues.technetworks.eu/view.php?id=1354>

IMPORTANT:

Please check-out our issue-tracker at <https://issues.technetworks.eu/> in the first month after a release! It common that we find new issues shortly after a public release which we fix very quickly afterwards. These fixed are then made available via so called "hot-fixes", which are minor updates for the current SMR only that are usually very small in size and can be installed during normal operational hours.



5. Device Support

In the next chapters we will inform you about the devices supported in this release.

5.1 Snom Devices

Various issues have been reported with the Snom phones. But even-though the Snom firmware is not part of this update, we will inform you about known issues and the best firmware to use. As all if this can have influence on the configuration or handling of these phones in the @COM.

5.1.1 Known issues

Therefore, we list various known issues here anyway...

The Snom D3, D7 and D7C key-modules can only be used on non-fully supported models. When used they cannot be programmed via the LMT as with "fully-supported" models.

[#4189](#) - From firmware v8.7.3 to v9, a phone no longer shows the name/number of waiting callers. You can still see a status line message that indicates there is a waiting caller, but not who is calling. In the past you saw who was calling at a separate line besides your current call.

[#4871](#) - The phone displays the error: 'Provisioning server failed'
This occasionally happens even when the provisioning server hasn't failed. You can simply erase the message by clicking on "Info" and delete the message in that menu

[#4667](#) - Certain models stops retrieving configuration from our server after a while, causing the display information to no longer be updated. A simple reboot will fix this for several weeks.

[#4196](#) - On attended transfer the call selected by default is not the last call placed on hold, but the newest call. This occurs on all firmware version higher than 8.9.3.101 and when the setting "xfer_dest_order_lifo" is set to "on" and the setting "call_join_xfer": is set to "off".

5.1.2 "Partial Support" for all newer Snom Desktop phones

Since SMR.1.3.1835-008-hf3 we provide "Partial Support" for all Snom Desktop phones that currently do not have "Full-support". The idea behind this "partial support" is that we can provision all of the latest Snom Desktop Phones without having to wait until we have built "Full-Support" for these models. There is a catch however, you cannot configure the function-keys or managing the firmware via the LMT. And you need to create the device using a "fully-supported" model that looks alike.

Currently all known Snom desktop phones are supported, except the old consumer 8xx series. Support for newest Snom 862, Snom 865, Snom 868 and Snom 895 models will follow soon once these phones become available.



5.1.3 Which firmware to use?

You **MUST** upgrade Snom phones to at least firmware version 8, as 7 is no longer provisioned properly.

We advise to use this firmware:

- For all Snom 300, 320, 360 and 370 models, use firmware 8.7.3.25.9.
- For all other “fully-supported” Snom models, we advise the use of firmware release v8.9.3.101.
- For all newer and partially supported Snom models such as the Snom D712, D717, D785, D305, D335, D385 and so on, we advise using the latest official Snom v10 firmware release.

5.1.4 Update v7 to v8 warning.

When upgrade from firmware 7 to 8 you **MUST** always first upgrade to firmware 7.3.30 by using the "snom3xx-7.3.30-SIP-bf.bin" firmware file. This is because this firmware version contains a new boot-loader (version 1.1.3u) which is not included in other files. After upgrading to this version, you can continue upgrading to other versions.

5.2 OpenStage Devices

OpenStage phones are still “Fully supported” but we will no longer built support for newer models or firmware. Also, not all features that can be provisioned are actually provisioned:

- A maximum of 2 lines can be handle at the same time making the phone not suitable for many receptionists and operators unless an ACD queue is used.
- No support by the @COM Connect for use with the PeterConnect Attendant.
- No NAT support, meaning these phones may work very poorly or not at all if place behind a NAT.
- No dial plan support, so users have to press a key or pick up the receiver to start the dial.
- No key programming via the phone menu itself. You can do so, but the programming will reset itself after a while.
- No interactive menus available yet as available on Unify PBX systems.
- No multiple language support for certain messages on the phone, only English is fully supported.
- The missed call LED cannot be turned off.
- No custom screensaver support.
- No custom ring tone support.
- No contacts photo support.
- Display information cannot be updated whilst the phone is in the Ringing or Connected state.

(For more info see issue: [#2129](#))

5.2.1 Which firmware to use?

We advise using the following firmware:

- OpenStage 15, use firmware V3 R3.40.0
- OpenStage 20/40/60, use firmware V3 R3.36.0
- OpenStage 80, use firmware V3 R3.17.0

IMPORTANT: Do not upgrade to higher firmware versions as the phone will no longer accept provisioning changes! If upgraded to a higher version, you will require the Unify DERT tool to downgrade the device.



5.3 Aastra/Mitel Devices

Various issues have been reported with the Aastra/Mitel phones. But even-though the Aastra/Mitel firmware is not part of this update, we will inform you about known issues and the best firmware to use. As all if this can have influence on the configuration or handling of these phones in the @COM.

5.3.1 Known issues

Therefore, we list various known issues here anyway...

#0000 – Certain Mitel 6867i model cannot be downgraded to firmware v4.3.x
This appears to happen in newer hardware releases only.
In this specific case use the latest official version 5 release available

[#3708](#) - DND can be activated on a Mitel 6863i phone itself.

[#4198](#) - Phone changes from headset to speaker mode when handling multiple calls.

[#4382](#) - Mitel expire timer issue (Rport 3581) causing slow registrations.

5.3.2 Which firmware to use?

We advise using the following firmware on @COM Business Manager v1.3 systems:

- All Aastra 6700 series models use firmware v3.3.1.4358
- All Mitel 6800 series models use firmware v4.3.0.2036

On @COM Business Manager v2 systems running SMR.2.162.234 of higher please use the latest v5 firmware.

NOTE:

Firmware v6 was still not available as public release whilst SMR.2.162.234 was being tested. As a result, we do not yet know if this firmware can be used or not, we will post an update on this subject as soon as we have more details.



6. Installation instructions

6.1 Specific installation notes & warnings for this release

The following **warnings (in red)** and notes **(in black)** apply for this release:

- **Always create a backup before attempting an update**
- **An update to at least SMR.2.1.185 is mandatory for this release**
- **Reboot of the system after update is required.**

When the update is scheduled, the system will automatically reboot after the installation of the update had completed. If the update is manually started, the administrator needs to manually click on the "Reboot" button to initiate the full reboot. Please close your browser window and wait until the server has restarted before reconnecting.
- **Mitel phones may reboot during the update.**

In this update we changed settings for Mitel phones that can cause an auto-reboot of the phone if the setting changes. So, be aware that some Mitel phones may reboot whilst installing this update.
- **A license is required for all LMT administrators since v2.87.127**

Check the amount of "Standard", "Advanced" and "Expert" level LMT administrators are present in the system, then check if there is a license already present on our license-server. We have checked all v2 systems for and provided the appropriate licensing already, but this check is just to be sure we didn't miss any.

6.2 How to activate this update

6.2.1 Manual update

Follow these instructions to manually install this update;

- Point your web browser to the URL for the LMT page of the @COM Business Manager. For example: <http://pbx.example.com/lmt>
- Login as "Expert" or "System Administrator" and click on the main menu item named "Service".
- Within the Service main menu select the Software submenu.
- Within the Software submenu select the option updates.
- An overview containing already applied updates will be shown to you. Just click on the Update button to proceed.
- Browse to the file containing the update, example; "update-SMR.2.87.127.tar.gz.pck"
- Press on the "Send" button to transmit the file to the @COM Business Manager. Based on the file size and the type of connection you are using this might take a while, after the file transfer a message will be displayed. When the message says "Release Information" the file-transfer was successful, any other message will explain itself.
- The release information message may also contain warnings that are of concern to this update. Please make notice of these warnings since they might inform you that the system will reboot thus dropping any active calls.



-
- If you're sure you want to continue, click on the "Update" button.
 - The update will now be started, and the progress can be seen by lines declaring what action has been taken. Please note that some actions might take quite a while, so no output for sometimes less than 5 minutes is normal.

When the update is successful a message will be displayed:

```
----- UPDATE SUCCESSFUL -----
```

A reboot will occur as soon as you click on the REBOOT button.
Close your browser window afterward and ignore any warning that may appear.
After reboot reconnect to the @COM Business Manager.

IMPORTANT:

Under no circumstance may the @COM Business Manager be turned off during the update process since this might lead to irreversible damage to the system.

!! DO NOT REBOOT THE SERVER IF ANY CHECKS HAVE FAILED!!

If the checks included in the update detect that the updated failed, contact TechNetworks Support immediately to have them rectify the issue. DO NOT REBOOT the server in the meantime as this may cause the system to become unreachable and then require on-site support to bring it back online.

NOTE: During every update a full backup is be made which allows TechNetworks engineers to restore the server fully or partially in case of the event this is required, this backup cannot be downloaded by system administrators.

6.2.2 Automatic Update

Automatic updating at specific time and date should work but was not tested.
It's therefor advised to perform a manual update, just to be sure.



6.3 How to enable Forced-on-PBX? (VAMO v2):

To enable the new Forced-on-PBX, simply follow these steps:

1. Set the correct "Forced-on-PBX inbound Prefix" at the route used for fixed/mobile extensions.
(For oneCentral AIR use: 8300)
2. Set the correct "Forced-on-PBX outbound Prefix" at the route used for fixed/mobile extensions.
(For oneCentral AIR use: 8900)
3. Add the (mobile) DID's for Fixed/Mobile extensions to the route.
4. Add a new Fixed/Mobile extension like you would add any other extension, but:
 - a) Set the "Fixed/Mobile number" in the correct format used by the provider. (No LCR is used!)
 - b) Set the "Forced-on-PBX via route" setting to the correct route.
 - c) Ensure you select "Send original Caller ID to Fixed/Mobile number" unless the user explicitly wants to see its own Caller ID on every incoming call. (Some users want this to determine if it is a private or office call.)

Inbound calls with the "Forced-on-PBX inbound Prefix" (8300) will now be directly routed to the DID defined at the fixed-mobile extension and calls to the Fixed-Mobile extension will be directly forwarded to the route set in the "Forced-on-PBX via route" setting and using prefix set by the "Forced-on-PBX outbound Prefix" (8900) at that route.

6.4 Convert deprecated "Forced-on-PBX custom-script" to fixed-mobile extensions

Conversion from the deprecated "*Forced-on-PBX: Capture and route outbound mobile calls*" custom-script to the new fixed-mobile extensions is done as follows:

1. Set the correct "Forced-on-PBX inbound Prefix" at the route used for fixed/mobile extensions.
(For oneCentral AIR use: 8300)
2. Set the correct "Forced-on-PBX outbound Prefix" at the route used for fixed/mobile extensions.
(For oneCentral AIR use: 8900)
3. Remove the old virtual extension(s) from the system.
Make sure to write down (copy/paste) all details first!
4. Add each fixed-mobile device as described in step 4 of chapter 6.3
5. When all extensions are moved over from virtual to fixed-mobile extension:
 - a) Remove the "Abbreviated number redirect context" in case it was only used for this purpose.
To find out if it was only used for this purpose, check if there are any other "Abbreviated numbers" that require to be accessed from external trunks.
 - b) Remove the "Forced-on-PBX: Capture and route outbound mobile calls" custom-script.

NOTE:
If you have many users with Forced-on-PBX to convert, please contact TechNetworks support. For a small fee they can convert all "Virtual extensions" to "Mobile Extensions" directly in the database.



7. Appendixes

7.1 What is a Software Maintenance Release (SMR)

A systems software configuration is the key element that determines the way a product behaves. This software configuration is also subject to change. That is the reason why every different software configuration is assigned its own numbering scheme called the Software Maintenance Release (SMR). The goal is to include every bit of software that is subject to change between different @COM products or within itself to be included in the SMR configuration. Customer specific configuration however is not included in the SMR.

One of the most important software configurations are the Asterisk® dial plan and the web pages of for example the Local Management Tool. These two components each have their individual numbering which changes with every change or bug fix. The total count of every change or bug fix determines the serial number part of the SMR and is preceded by the main release number and the letters "SMR".

For example, SMR.2.87.127 stands for:

- **SMR** – Software Maintenance Release
- **2** – Indicating the main release v2 used in all @COM BM v2 systems. (major level)
- **87** – Indicating the number of feature-requests and changes (minor level).
- **127** – Indicating the number of bug-fixes and minor changes with full backwards compatibility. (patch-level)

For more information about how the numbering of software releases work, check www.semver.org

7.2 Full Change Log

The full change log can be viewed online at https://issues.technetworks.eu/changelog_page.php.





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