

SCOOP *Manager*

Codecs manager



User manual

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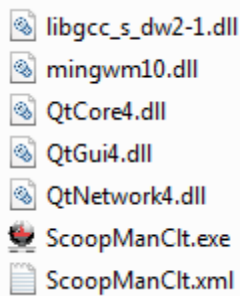
Scoop Manager is a professional solution that allows you to manage your AETA codecs from a single screen. You can easily see and configure all your codecs in order to manage your calls on various networks (IP/ISDN/Phone/Mobile).



1. Installation and activation of Scoop Manager

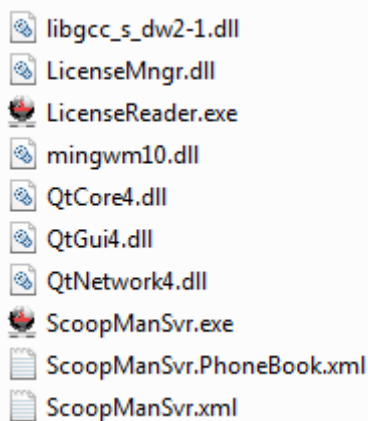
1.1. *Installation of the client application*

Copy the content of the "client" and "common" directories of the CD-ROM in the installation directory of the client application on the target computer A, then update the "ScoopManClt.xml" configuration file with the IP address of the target computer B that will host the server application.



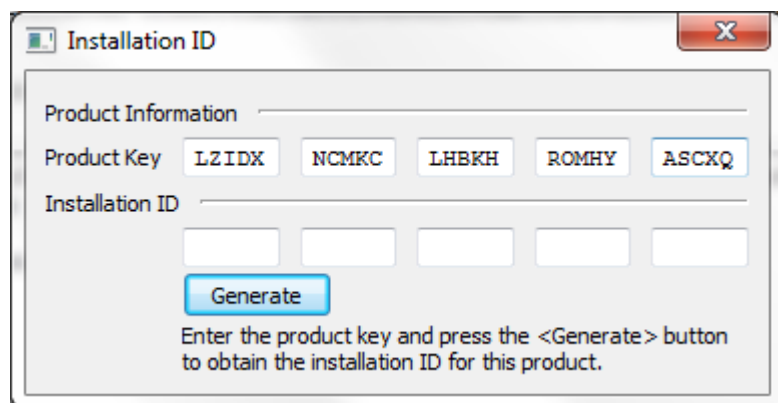
1.2. *Installation of the server application*

Copy the content of the "server" and "common" directories of the CD-ROM in the installation directory of the server application on the target computer B.



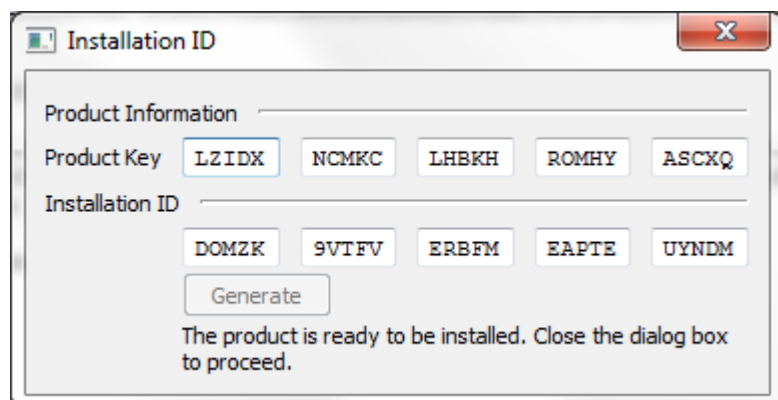
1.3. Activation of the server application

Launch the server application, then enter your product key.



The "Installation ID" dialog box contains a "Product Information" section with five buttons for product keys: LZIDX, NCMKC, LHBKH, ROMHY, and ASCXQ. Below these is an "Installation ID" section with five empty text boxes and a "Generate" button. A message at the bottom states: "Enter the product key and press the <Generate> button to obtain the installation ID for this product."

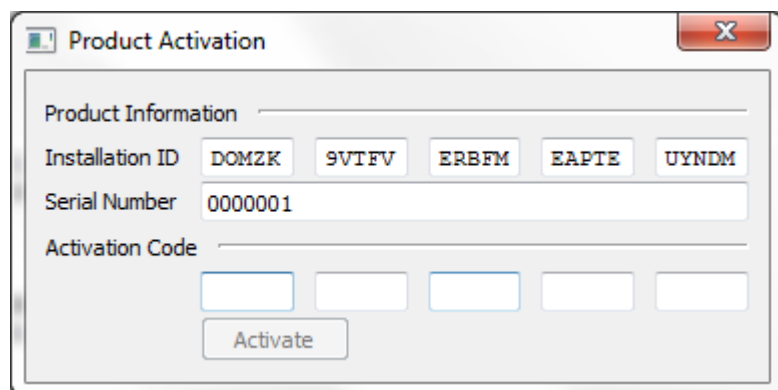
Click on the "Generate" button to get the installation identifier for the target computer B and send this identifier by email to the following address: "support@aeta-audio.com".



The "Installation ID" dialog box shows the same product key buttons. The "Installation ID" section now displays five buttons with generated identifiers: DOMZK, 9VTFV, ERBFM, EAPTE, and UYNDM. The "Generate" button is disabled. A message at the bottom states: "The product is ready to be installed. Close the dialog box to proceed."

Close this dialog box and the following ones, then close the application.

Once you have received an activation code for the target computer B, launch the server application, enter the activation code and click the "Activate" button.



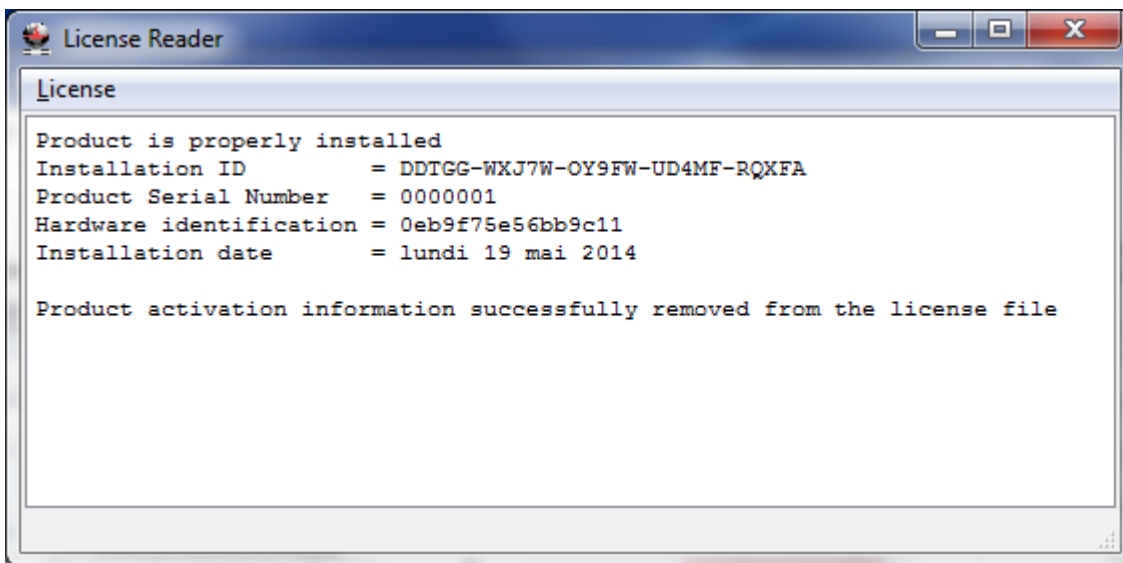
The "Product Activation" dialog box contains a "Product Information" section with five buttons for installation IDs: DOMZK, 9VTFV, ERBFM, EAPTE, and UYNDM. Below these is a "Serial Number" field with the value "0000001" and an "Activation Code" section with five empty text boxes and an "Activate" button.

1.4. Modification of the software options of the server application

Here is the procedure to follow to enter a new activation code to get a perpetual license or change the software options of the server application.

In the installation directory of the server application, make a backup copy of the "ScoopManSvr.lic" file; so if something goes wrong while executing this procedure, you can always go back and continue to use your current license. (See Note 2 below in this topic.)

In this directory, launch the "License Reader.exe" tool, then remove the information related to the product activation stored in the "ScoopManSvr.lic" file by selecting "Shrink License ..." from the "License" menu. For instance:



Exit the "License Reader.exe" tool.

Exit and restart the server application, enter your new activation code and click the "Activate" button.

Note 1: you can safely execute this procedure and reactivate the application server even if some of your codecs are part of active connections.

Note 2: if the procedure fails, exit the server application, restore the original "ScoopManSvr.lic" file and restart the application.

2. Presentation – Getting started

2.1. Server application

The server application manages a pool of AETA codec via an Ethernet LAN, using a proprietary control protocol.

This application allows the following: administration of the managed codecs (device list, naming of the codecs, management rules of the phone number prefix for the ISDN network for each device, list of the applicable encoding presets), monitoring of the status of the managed codecs (connected or not), managing a system wide phone book.

The configuration parameters of the server application are stored in the "ScoopManSvr.xml" file and must be edited manually. A server application restart is required after any changes in this file. The connectivity settings of the server application are stored at the "server" node level. The "listeningPort" parameter defines the listening TCP/IP port that the client applications will use to connect to the server application; the "password" parameter specifies the password to be used by a user from a client application to perform administrative tasks.

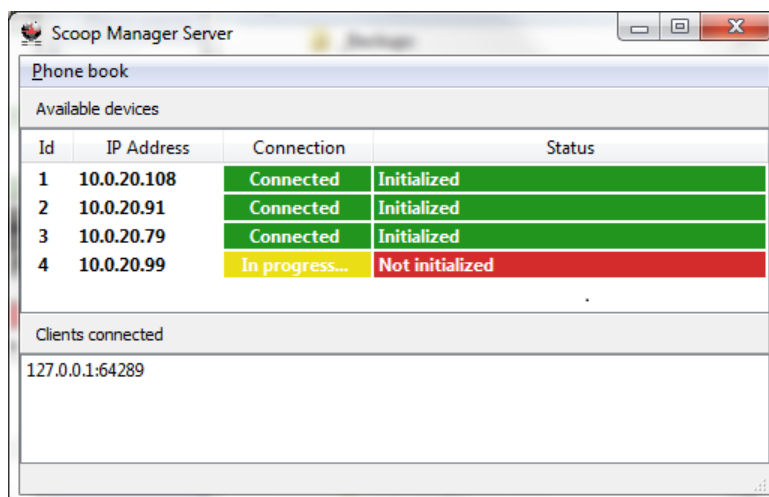
```
<settings version="1">
  <server password="admin" listeningPort="7000"/>
```

The various other configuration parameters of the server application are detailed later in this document.

2.1.1. Graphical user interface of the server application

The GUI of the server application offers the following features:

- Display of the communication status with the client applications (connected or not).
- Display of the status of the managed codecs (connected or not).
- Ability to import / export the system wide phone book from / to a local directory, or the local network, using a standard dialog box.



The English is the default language for the GUI. However, at application startup, the GUI may automatically switch to French or German depending on the system's locale settings. It is always possible to override the system's locale settings by passing the target language as an argument to the "ScoopManSvr.exe" executable file: "en" for English, "fr" for French and "de" for German, in accordance with the ISO 639 standard, which defines the codes for the representation of names for language and language groups.

Although the application server manages the system wide phone book, said phone book content is actually edited using the GUI of the client application. In other words, the server application has no dedicated GUI to add, modify and delete contacts in this phone book.

Note: the "ScoopManSvr.PhoneBook.xml" file that stores the phone book is manually editable or can be overwritten by another phone book file, any changes require a server application restart to be taken into account.

2.1.2. Codec pool management

The list of codecs managed by the server application, the names associated with the sub-codecs n°1 and n°2 of these devices, the individual management rules (i.e. for each codec) of the phone number prefix in case a call is initiated or received on a ISDN network, are stored in the "ScoopManSvr.xml" file that must be edited manually. A server application restart is required after any changes in this file.

```
<knownDevices>
  <device id="1" address="10.0.20.86" port="7001" datagramPort="3000">
    <codec index="1" name="CODEC:1"/>
    <codec index="2" name="CODEC:2"/>
    <isdnPrefixRules incomingCall="auto" outgoingCall=""/>
  </device>
```

The configuration parameters for each of the managed codec are stored at a "device" node level, under the "knownDevices" node. The "id" parameter is used to uniquely identify a codec in the file "ScoopManSvr.xml"; the "address" parameter defines the TCP/IP address of the codec, and the parameter "port" defines the TCP/IP port on which it is reachable at this address (7001, by default); the "datagramPort" parameter indicates to which UDP/IP port of the server application the codec will send its audio levels, each codec must use a different port in the "ScoopManSvr.xml" file. (See also **"Appendix A: Case of a MultiScoop system"**)

The name associated with the sub-codecs n°1 and n°2 is stored at a "codec" node level, under the "device" node. The "index" parameter specifies the sub-codec number (1 or 2); the "name" parameter defines the name to associate with the sub-codec and that will be displayed by the client application.

2.1.3. Coding presets management

The list of networks (ISDN, Ethernet, mobile, etc.) supported by a codec managed by the server application, that will be available on the client application, is gathered dynamically via the control protocol; this list is not stored in a file.

The server application holds the list of the coding presets that will be available on the client application. These are system wide presets, applicable to each of the codecs managed by the server application, provided that the coding preset is suitable for the target codec. The coding presets are stored in the "ScoopManSvr.xml" file and must be manually edited. A server application restart is required after any changes in this file.

```
<codingPresets>
  <preset name="G.711" algorithm="G711" audioMode="" sampleRate="" bitRate="" errorCorrection=""/>
  <preset name="L16" algorithm="L16" audioMode="mono" sampleRate="48000" bitRate="" errorCorrection=""/>
  <preset name="MPEG2" algorithm="MPA2P" audioMode="stereo" sampleRate="48000" bitRate="128000" errorCorrection=""/>
```

The coding preset settings are stored at a "preset" node level, under the "codingPresets" node. The "name" parameter defines the name of a coding preset as it will be displayed on the client application; the "algorithm" parameter indicates the coding algorithm for that coding preset (see table below); the "audioMode" ("mono",

"stereo", "dmono" or "jstereo"), "sampleRate" (sampling rate, in Hz), "bitRate" (bits per second) and "errorCorrection" parameters define the settings of the coding for this preset.

The following table shows the suitable values for the "algorithm" parameter based on the coding algorithm and whether or not the other parameters are required for this algorithm:

Coding algorithm	algorithm	audioMode	sampleRate	bitRate	errorCorrection
Celp	CELP				YES
G.711 a-law (IP)	PCMA				
G.711 u-law (IP)	PCMU				
G.711 (ISDN)	G711				
G.722 (IP)	G722				
G.722 SRT (ISDN)	G722-SRT				
G.722 H.221 (ISDN)	G722-H221				
G.722 H.242 (ISDN)	G722-H242				
4SB ADPCM	4SB	YES	YES	YES	
MPEG L2 J.52	MPA2	YES	YES	YES	YES
MPEG L2	MPA2P	YES	YES	YES	
MPEG L3	MPA3	YES	YES	YES	
AAC-LC ⁽¹⁾	AAC-lbr	YES	YES	YES	
AAC-HE ⁽¹⁾	AAC-he	YES	YES	YES	
AAC-HEv2 ⁽¹⁾	AAC-hev2	YES	YES	YES	
AAC-LC LOAS (ISDN) ⁽²⁾	AAC-lbr-loas	YES	YES	YES	
AAC-HE LOAS (ISDN) ⁽²⁾	AAC-he-loas	YES	YES	YES	
AAC-HEv2 LOAS (ISDN) ⁽²⁾	AAC-hev2-loas	YES	YES	YES	
AAC-LD	AAC-ld	YES	YES	YES	
AAC-ELD	AAC-eld	YES	YES	YES	
Voix Mobile	GSM				
POTS Phone	Phone				
Opus	OPUS	YES	YES	YES	
L16	L16	YES	YES		
L20	L20	YES	YES		
L24	L24	YES	YES		

⁽¹⁾ ADTS (Audio Data Transport Stream) framing for ISDN networks.

⁽²⁾ LOAS (Low Overhead Audio Stream) framing.

2.1.4. Call history management

The server application holds the call history that will be available at the client application level. This history is global to the system, that is to say it chronologically lists all the calls initiated or received by the whole codecs managed by the server application. The call history is stored in the "ScoopManSvr.CallHistory.xml" file and is not intended to be manually edited. The format of this file and the nature of the information stored in this file are the property of AETA and are not intended to be documented.

The depth of the call history can be configured in the "ScoopManSvr.xml" file which requires to be manually edited. A server application restart is required after any changes in this file.

```
<applicationOptions>
  <callHistory maxCapacity="100"/>
</applicationOptions>
```

The "maxCapacity" parameter defines the maximum number of calls that will be stored in the "ScoopManSvr.CallHistory.xml" file. The default depth for the call history is set to 100 calls.

2.1.5. Server application maintenance

The log file generation can be configured in the "ScoopManSvr.xml" file which requires to be manually edited. A server application restart is required after any changes in this file. The format of this file and the nature of the information stored in this log file are the property of AETA and are not intended to be documented.

```
<applicationOptions>
  <logger logMaxSize="512" logMaxFiles="20"/>
</applicationOptions>
```

The "logMaxSize" parameter specifies the maximum size of a log file, in kilobytes; the "logMaxFiles" parameter defines the number of log files to store on the hard drive. In the above example, the server application keeps the last 20 log files, each having a maximum size of 512 kilobytes.

2.2. Client application

The client application connects to the server application via an Ethernet LAN, using a proprietary communication protocol.

This application allows the supervision of the codecs (i.e. current status of the codec, their configuration for the outgoing calls, the current status of any active call, the audio level, etc.), the configuration and management of the calls (i.e. configure the network / coding algorithm for the outgoing calls, initiate an outgoing call, release an incoming or outgoing call), and the management of the system wide phone book.

The English is the default language for the GUI. However, at application startup, the GUI may automatically switch to French or German depending on the system's locale settings. It is always possible to override the system's locale settings by passing the target language as an argument to the "ScoopManClt.exe" executable file: "en" for English, "fr" for French and "de" for German, in accordance with the ISO 639 standard, which defines the codes for the representation of names for language and language groups.

Note: The GUI of the client application is not intended to be operated via a touch screen; a keyboard and a mouse are mandatory.

2.2.1. Client application management

The client application settings are stored in the "ScoopManClt.xml" file which must be manually edited. The client application must be restarted after any changes in this file.

```
<settings version="1">
  <client logMaxSize="512" logMaxFiles="20"/>
  <server address="127.0.0.1" port="7000" datagramPort="7001"/>
  <devices columnCount="4" widgetOptions="0x01">
    <device id="1" widgetOptions="0x03" incomingCallSound="" hangupCallSound=""/>
    <device id="2" incomingCallSound="" hangupCallSound=""/>
    <device id="3" incomingCallSound="" hangupCallSound=""/>
  </devices>
</settings>
```

Within the "client" node, the "logMaxSize" parameter defines the maximum size of a log file in kilobytes, the "logMaxFiles" parameters defines the number of log files to keep on the hard drive.

Within the "server" node, the "address" and "port" parameters define respectively the TCP/IP address and the listening port of the server application to connect to; the "datagramPort" parameter indicates on which UDP/IP port the client application wants to receive the audio levels.

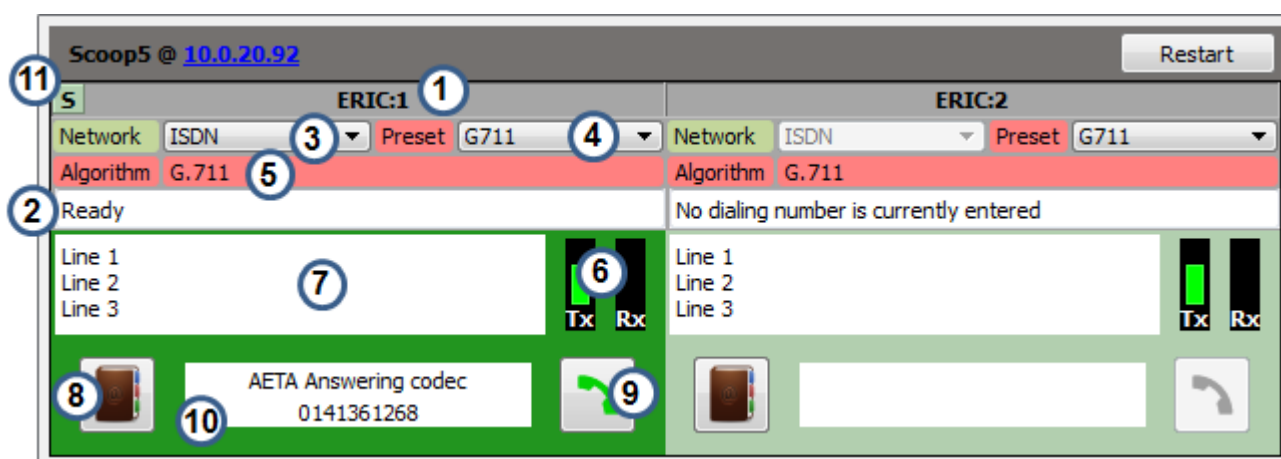
The configuration of the list of supervised codecs and their disposition on the GUI of the client application is configured in the "devices" node. The "columnCount" parameter indicates how many codecs will be displayed horizontally. The "device" nodes are references to the codecs to display. For each "device" node, the "id" parameter identifies one of the codecs registered within the file "ScoopManSvr.xml" of the server application; the "incomingCallSound" and "hangupCallSound" parameters allow to configure two wave files (the only audio format supported by the client application) that will be played respectively when the underlying codec receives a call and when an active call is released.

The optional "widgetOptions" parameter of the "devices" and "device" nodes allows customizing the behavior and graphical rendering of the client application. This setting is detailed later in this document.

2.2.2. Visual representation of a Scoop5 in the graphical user interface

Each codec is represented by a frame which includes two insets aligned horizontally. This frame has a button in its upper right corner which allows restarting the codec; the user is prompted before proceeding and needs to provide a password to complete the action (one password is defined for the whole system; it is common to all the codecs). The left inset represents the sub-codec n°1 of the codec; the right inset represents the sub-codec n°2 of the codec. The graphical rendering of the right inset depends on the single codec / double codec mode configuration of the supervised device.

Graphical rendering of a codec configured in double codec mode



- (1) This area allows identifying the sub-codec associated with the inset, the text displayed in this area is configured by the server application. The server application does not limit the number of characters to name the sub-codecs; therefore, the text displayed in this area can be a truncated version of the actual name associated with the sub-codec.
- (2) The background color of the inset is used to indicate the general state of the codec, as well as the audio connection status of the sub-codec associated with this inset. A non-editable text field, under the control of

the client application, completes this information and explicitly details the status of the codec or any error messages.

A set of six colors are used for the inset:

Grey	Connection lost with the codec or the codec is not properly configured.
Light green	Codec is properly configured, but no outgoing call number has been entered yet.
Dark green	Codec is properly configured, an outgoing call number has been entered and the codec is ready to establish an audio connection.
Yellow	Call is proceeding (incoming / outgoing call).
Red	Call is established (incoming / outgoing call).
Orange	Call is being released (incoming / outgoing call).

- (3, 4) The inset allows a handy management of the configuration of its associated sub-codec, while it is idle. Two combo boxes are used respectively to configure the network (ISDN, Ethernet, mobile, etc.) and to select the coding algorithm for the outgoing calls (coding presets, managed by the server application). When an audio connection is active, these combo boxes are greyed out, to prevent from modifying the sub-codec configuration during the call.

Whenever the configuration of a sub-codec is changed from the front panel of the codec or from the web pages of the codec, the fields (3) and (4) are automatically refreshed. However, if the newly applied coding parameters do not match a coding preset, the field (4) will display a blank text while the field (5) will show all the coding parameters.

The configuration information of the sub-codec is always visible on the inset, whatever the audio connection status of the sub-codec.

The client application manages the consistency of the networks and the coding algorithms listed in the fields (3) and (4) based on the options supported by the sub-codec attached to the inset. In other words, the networks proposed by the field (3) are those actually supported by the target device, the coding presets proposed by the field (4) are valid and suitable presets for this device.

- (5) When the sub-codec is idle, this field displays the coding parameters selected for an outgoing call (coding algorithm, audio mode, sampling rate, bit rate, etc.). When an audio connection is established and as soon as the related information is available, this field shows the "negotiated" coding parameters.
- (6) Transmit and receive audio levels of the sub-codec are displayed within the inset.
- (7) The inset provides the user with a multi-line text field; that field is freely editable by the user, whatever the sub-codec audio connection status. Moreover, this field supports standard copy / paste operations from the clipboard.
- (8, 10) When the sub-codec is idle, the button (8) of the inset opens the phone book, allowing the user: to manage the contacts of the system wide phone book; to dial a remote number and fill in the field (10); to select a remote number from the phone book and fill in the field (10). The features related to the phone book are detailed in the chapter 2.3, later in this document.

The field (10) is used to display the name of a caller and its associated remote number. While the sub-codec is idle, the information of this field indicates the parameters that will be used to initiate an outgoing call; during an active audio connection, the information corresponds to the parameters of the outgoing call or the incoming call in progress.

A right click on the field (10) shows a contextual menu that allows erasing the content of this field. The content of the field (10) is not memorized, when the client application restarts, the field is blank.

In the case of an ISDN call, only the first number is displayed in the field (10) and is subject to the management rules for the phone number prefix. Indeed, depending on the coding algorithm selected for an ISDN call, up to four numbers may be required.

In addition, this field supports the standard copy operation of the clipboard.

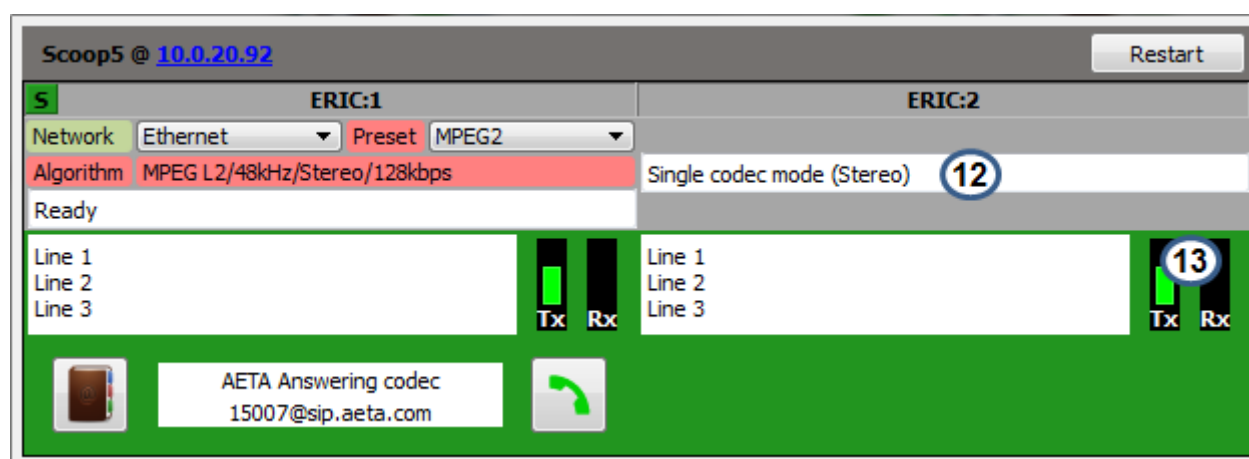
- (9) While the sub-codec is idle, whenever the sub-codec is properly configured and the remote number for an outgoing call is properly entered in the field (10) and the codec is ready to initiate an audio connection, the button (9) of the inset is used to initiate that outgoing call. During an active audio connection, this very same button is used to release the call.
- (11) The indicator (11) shows the current SIP registration status of the codec.

A set of four colors is used for this indicator:

- Grey The SIP account is disabled for the codec.
- Dark Green The codec is properly registered and the network configured by the field (3) is the network used for the SIP calls.
- Light Green The codec is properly registered but the network configured by the control (3) is not the network used for this SIP calls.
- Red The codec fails to register on the SIP server.

Graphical rendering of a codec configured in single codec mode

The following figure shows the graphical rendering of a codec configured in single codec mode when the "automatic hide" option associated with the second sub-codec is disabled. (See "Customizing the behavior and the graphical rendering of the client application" later in this document)



- (12) This area of the second inset (read only field) indicates to the user that the codec is configured in single codec mode. In addition, the current audio mode is shown between brackets.
- (13) Transmit and receive audio levels of the codec are displayed within the inset. However, if the current audio mode is "Mono", these bar graphs will be hidden for the second inset (i.e. second sub-codec).

2.2.3. Customizing the behavior and graphical rendering of the client application

By default, when the network is configured to ISDN, the client application will automatically switch the codec in double codec mode if the configured algorithm allows it (G711, G722 SRT, G722 H.221 and G722 H.242). However, it is possible to modify the "ScoopManClt.xml" configuration file to change this default behavior.

Simply add the "widgetOptions" parameter in the "devices" node to define a global behavior for the whole managed codecs, or at the "device" nodes level to define the behavior of a given codec. The "widgetOptions" parameter value is made of a set of bits:

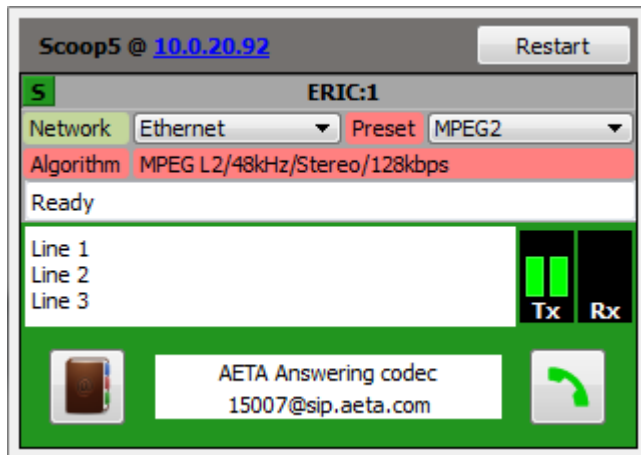
- Set the bit number 0 to 1 to automatically hide the inset associated with the second sub-codec when the codec is configured in single codec mode.
- Set the bit number 1 to 1 to enable the automatic switching of the single / double codec mode when the algorithm allows it.
- When the codec is not connected or not initialized yet, respectively set the bit number 2 to 0 to display the frame in single codec mode, to 1 to display the frame in double codec mode.
- Set the bit number 3 to 1 to lock the inset associated with the first sub-codec.
- Set the bit number 4 to 1 to lock the inset associated with the second sub-codec.
- Set the bit number 5 to 1 to lock the background color of the inset (requires bit number 3 and/or bit number 4 to be set to 1).
- Set the bit number 6 to 1 to enable direct editing of the (10) speed dial number.
- Set the bit number 7 to 1 to display the call history page by default when the (8) phone book is opened.

When the "widgetOptions" is omitted within the "ScoopManClt.xml" configuration file, the global behavior corresponds to "widgetOptions = 0x03".

Example for a pool made of three Scoop5:

```
<settings version="1">
  <client logMaxSize="512" logMaxFiles="20"/>
  <server address="127.0.0.1" port="7000" datagramPort="7001"/>
  <devices columnCount="4" widgetOptions="0x01">1
    <device id="1" widgetOptions="0x03"2 incomingCallSound="" hangupCallSound=""/>
    <device id="2" incomingCallSound="" hangupCallSound=""/>
    <device id="3" incomingCallSound="" hangupCallSound=""/>
  </devices>
</settings>
```

(1) The global behavior for all the Scoop5 is to automatically hide the inset associated with the second sub-codec when the Scoop5 is configured in single codec mode and to disable the automatic switching of the single / double codec mode, (2) except for the first Scoop5 for which the automatic switching of the single / double codec mode is enabled.



The opposite figure shows the graphical rendering of a codec configured in single codec mode when the option to automatically hide the inset associated with the second sub-codec is enabled.

Additional graphical customizations

It is also possible to customize the font size and vertical size of certain UI controls as well, by modifying the "devices" node.

```
<settings version="1">
  <client logMaxSize="512" logMaxFiles="20"/>
  <server address="127.0.0.1" port="7000" datagramPort="7001"/>
  <devices columnCount="3" widgetOptions="0xC1" infoPointSize="14" textPointSize="12" textLineCount="2">
    <device id="1" incomingCallSound="" hangupCallSound=""/>
  </devices>
</settings>
```

- The "infoPointSize" defines the font size of the fields (1) codec identification and (10) call number. The font size is 8 points, by default if this setting is omitted.
- The "textPointSize" defines the font size of the (7) free text area. The font size is 8 points, by default if this setting is omitted.
- The "textLineCount" parameter defines the vertical size of the (7) free text area in number of lines.

Managing the multi-line text field persistency

By default, the text entered in the field (7) is not memorized, therefore when the client application is restarted, the field is left blank. It is possible to modify this default behavior by adding a "textPersistence" parameter in the "devices" node and set it to either "local" (the persistency is managed locally by the client application) or "server" (the persistency is managed at server side and every single client application connected to the server is automatically updated whenever a text field is changed), for instance:

```
<settings version="1">
  <client logMaxSize="512" logMaxFiles="20"/>
  <server address="127.0.0.1" port="7000" datagramPort="7001"/>
  <devices columnCount="3" widgetOptions="0xC1" textPersistence="local">
    <device id="1" incomingCallSound="" hangupCallSound=""/>
  </devices>
</settings>
```

2.3. Phone book management

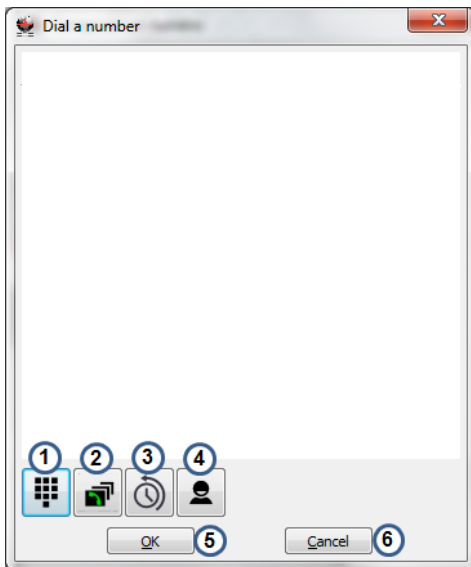
The management of the phone book is only possible when the codec is idle (from the inset associated with the sub-codec); simply click the button (9) of an inset, a dialog box opens, which allows to:

- Create, edit and delete contacts in the phone book. A contact is defined as: a contact name associated with a series of remote numbers, each remote number is associated with a given type of network (for ISDN, note that up to four numbers may be necessary). For example, a contact can be configured to initiate an outgoing call to a remote party either using an ISDN or a mobile network. The main advantage of linking a remote number to a given type of network, for which this remote number is applicable, is the ability to apply a filter on the entire phone book and therefore to list the contacts that are relevant for this network type.
- Dial a remote number for the codec (according to the coding algorithm selected for an ISDN call, up to four numbers may be necessary). It is the remote number that will be displayed in field (10) of the inset; no contact name will be displayed.
- Select a remote number from the contact list. The contact list is filtered out against the current configuration of the codec, only the relevant contacts are returned. The name of the contact and the associated remote number will be displayed in the field (10) of the inset (in the case of an ISDN call, only the first number is displayed).
- Browse the history of the outgoing and incoming calls; select an entry and call the contact back using the same network configuration and the same coding parameters previously negotiated for that call.

To simplify operations in this dialog box, the standard copy / paste operations of the clipboard are available.

2.3.1. Graphical user interface of the phone book

Here is the general aspect of the phone book GUI; the layout of the UI controls remains the same in all the phone book management modes. Four buttons (1, 2, 3 and 4) allow the user to switch between the four management modes of the phone book: dial a remote number, select a contact, browse the call history and manage the contact list. The white frame in the following figure will display UI controls dedicated to the current management mode. The buttons (4 and 5) are respectively used to validate an action (with a confirmation window, whenever required) and to cancel an action.

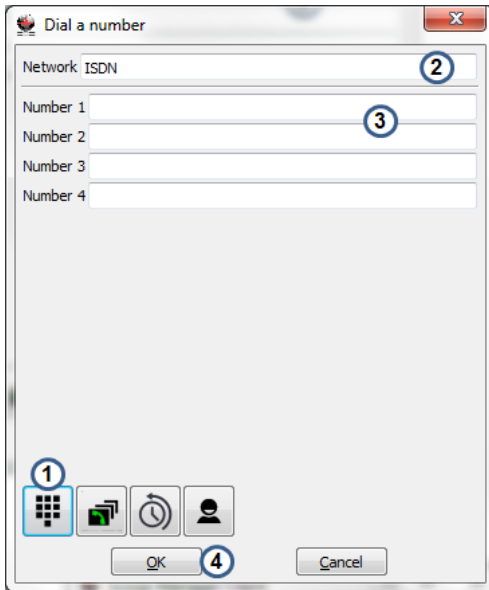


- (1) Press this button to enter the "Dial a remote number" mode (detailed in Chapter 2.3.2).
- (2) Press this button to enter the "Select a remote number from the contact list" mode (detailed in Chapter 2.3.3).
- (3) Press this button to enter the "Browse the call history" mode (detailed in Chapter 2.3.4)
- (4) Press this button to enter the "Contact list management" mode (detailed in chapters 2.3.5 and 2.3.6).

The highlighted button, the button (1) in the opposite image, indicates the user which management mode is currently active.

2.3.2. Dial a remote number

The following figure shows the general aspect of the window that allows the user to dial a remote number.



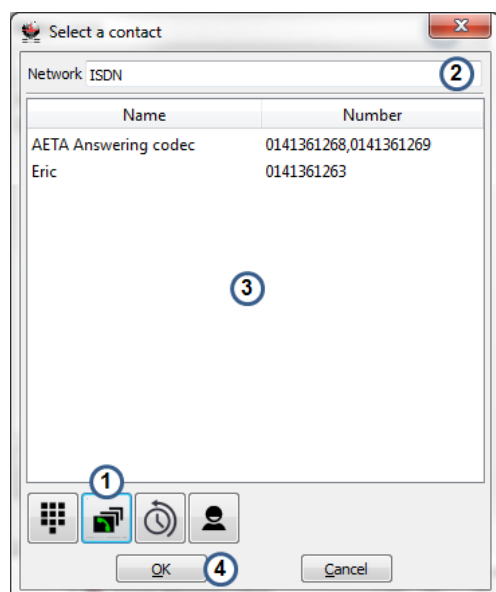
- (1) The button is highlighted to indicate that the current management mode is "Dial a remote number."
- (2) This text box (read only) reminds the user of the current configured network for the sub-codec associated with the inset (ISDN, Ethernet, Mobile, etc.).
- (3) Depending on the coding algorithm selected for this network, the user may enter up to four telephone numbers (in the case of a 256 kbps coding algorithm for an ISDN network).

The format of the remote numbers is detailed in chapter 2.3.7.

- (4) Press this button to confirm the settings that will be used for the outgoing calls.

2.3.3. Select a remote number from the contact list

The following figure shows the general aspect of the window that allows the user to select a remote number from the contact list.



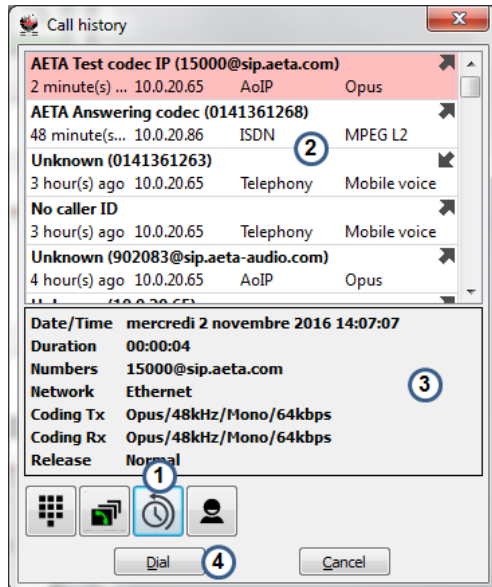
- (1) The button is highlighted to indicate that the current management mode is "Select a remote number from the contact list."
- (2) This text box (read only) reminds the user of the current configured network for the sub-codec associated with the inset (ISDN, Ethernet, Mobile, etc.)
- (3) The user selects a contact in the list before clicking the button (4); contacts are sorted alphabetically.

The contacts shown in (3) are a subset of the contact list of the phone book; only the contacts that are suitable with the currently selected coding algorithm and network are listed. The displayed remote number is the one associated with the network type for this contact (the first number is chosen if several numbers are configured, in the case of a 256 kbps coding algorithm for an ISDN network, for instance).

- (4) Press this button to confirm the settings that will be used for the outgoing calls.

2.3.4. Browse the call history

The following figure shows the general aspect of the window that allows the user to browse the call history.



- (1) The button is highlighted to indicate that the current management mode is "Browse the call history."
- (2) The user selects an entry from the call history to display the details of this call in the area (3) and allows calling the contact back with the button (4).

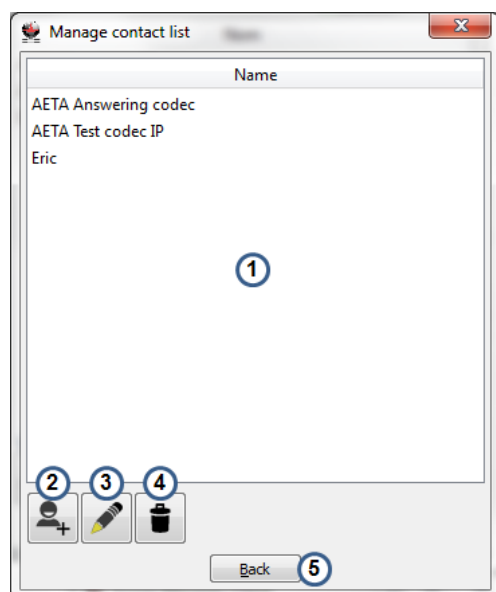
The first line shows the caller's name followed by the call number. The second line shows the codec concerned by the call, the network type for this call and the coding algorithm negotiated during the call.

- (3) This text box (read only) details the parameters of the selected call in the history.
- (4) This button allows calling the contact with the sub-codec associated with the inset, using the same network and the same coding parameters previously negotiated for that call.

At the end of the call, the network configuration and the coding algorithm selection for the outgoing calls for that inset are restored.

2.3.5. Contact list management

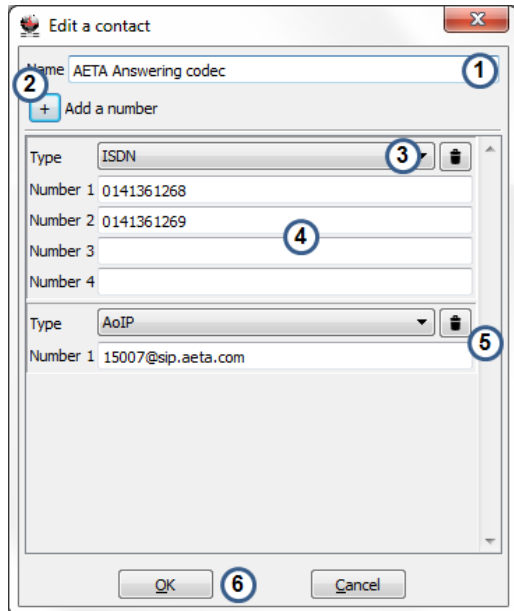
The following figure shows the general aspect of the window that allows the user to manage the phone book, that is to say the whole contacts.



- (1) The user selects a contact in the list before clicking on the button (3) or (4); contacts are sorted alphabetically.
- (2) This button allows the creation of a new contact, as detailed in section 2.3.6.
- (3) After selecting a contact in the list (1), this button allows you to edit the associated parameters, as detailed in section 2.3.6.
- (4) After selecting a contact in the list (1), this button will permanently delete the contact from the phone book; the user is prompted to confirm the action before proceeding.
- (5) This button allows the user to return to the main window of the phone book.

2.3.6. Create / Edit a contact

The following figure shows the general aspect of the window that allows the user to modify the parameters associated with a contact. For each contact, the user can define a remote number depending on the underlying network type: "ISDN", "AoIP" (Mobile network or Ethernet network in data mode) and "Telephony" (POTS or mobile voice mode). In other words, a contact can be configured to allow calling the same remote party via an ISDN network or via a mobile network.



- (1) This text box allows you to enter the name to associate with this contact.
- (2) Click this button to add a new entry to the list of remote numbers for this contact. Up to three different call settings are available for a contact: "ISDN", "AoIP" and "Telephony".

For a given call setting, the user can:

- (3) Select the type of the call: "ISDN", "AoIP" or "Telephony".
- (4) Associate up to four remote numbers for this setting (in the case of a 256 kbps coding algorithm for an ISDN network).

The format of the call numbers is detailed in chapter 2.3.7.

- (5) Delete an existing call setting.
- (6) Press this button to validate the changes for this contact, the user is prompted to confirm before proceeding. An error message will be displayed if the name entered in the field (1) is already used by another contact in the phone book.

2.3.7. Remote number formats

Here is the format of the remote numbers depending on call setting type:

ISDN	ISDN network
	number (e.g. "0141361268")
	number*sub-address (e.g. "0141361268*1234")
AoIP	Ethernet network or Mobile network in data mode
	address (e.g. "192.168.100.12")
	address:port
	name@address
	name@address:port
	name@hostname (e.g. "1500@sip.aeta-audio.com")
	name@hostname:port (e.g. "1500@sip.aeta-audio.com:5060")
Telephony	POTS or Mobile network in voice mode
	number (e.g. "+33141361200")

Description:

number Sequence of digits, possibly preceded by the « + ».

sub-address Up to four numbers

address An IPv4 address format.

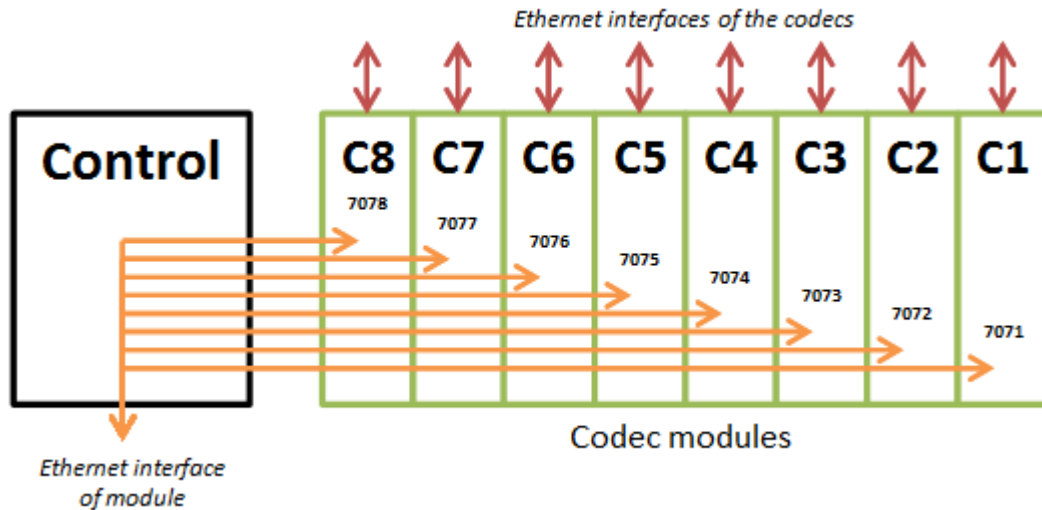
port Number between 1 and 65535.

name, hostname Alphanumeric characters, characters "-", ".", "_" and "~".

Note: The phone book manager does not validate the format of remote numbers entered by the user.

3. Appendix A: Case of a MultiScoop system

The "Control" module of a MultiScoop system acts as a gateway for accessing the "Codec" modules installed in the MultiScoop chassis. This capability applies both to the embedded html pages and to the control protocol used by the Scoop Manager software. However, the configuration of the "ScoopManSvr.xml" file requires special attention when the Scoop Manager software controls the "Codec" modules of a MultiScoop system through its "Control" module.



In the following example, the IP address of the "Control" module is "10.0.20.81" and the Scoop Manager remote controls the "Codec" modules located in slots # 1 and # 2 of the MultiScoop chassis.

```
<knownDevices>
  <device id="1" address="10.0.20.81" port="7071" datagramPort="1">
    <codec index="1" name="CODEC1:1"/>
    <codec index="2" name="CODEC1:2"/>
    <isdnPrefixRules incomingCall="auto" outgoingCall=""/>
  </device>
  <device id="2" address="10.0.20.81" port="7072" datagramPort="1">
    <codec index="1" name="CODEC2:1"/>
    <codec index="2" name="CODEC2:2"/>
    <isdnPrefixRules incomingCall="auto" outgoingCall=""/>
  </device>
</knownDevices>
```

For every single "device" node related to the "Codec" module of a given MultiScoop system:

- The "address" parameter defines the IP address of the "Control" module.
- The "port" parameter, which format is "707[1..8]", simultaneously defines:
 - The number of a slot in the MultiScoop chassis,
 - The TCP/IP port on which the "Codec" module of this slot can be reached.
- The "datagramPort" parameter must be set to "1" (i.e. the transmission of audio levels from the "Codec" module through the "Control" module and to the Scoop Manager server application will be carried out over TCP/IP).



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