

LCOS 10.34

Addendum



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1 Addendum to LCOS version 10.34

This document describes the changes and enhancements in LCOS version 10.34 since the previous version.

2 Voice over IP – VoIP

2.1 Dynamic SIP lines

From LCOS 10.34 it is possible to set up dynamic SIP lines. The configuration is found under **Voice call manager > Lines** by clicking the button **Dynamic SIP line**.

The screenshot shows the 'SIP lines' configuration page. It contains the following elements:

- SIP lines** section:
 - Text: "Here, you may configure lines for public SIP providers for which the router registers itself. Outgoing calls may be made via Call Router on these lines."
 - Buttons: "SIP lines..." and "Dynamic SIP-Line..."
 - Text: "The SIP mapping table can be used to specify internal and external numbers for trunk and gateway SIP lines."
 - Button: "SIP mapping..."
 - Text: "In the SIP PBX lines table you can define upstream SIP phone systems (PBX), for which all local users with a PBX-adequate domain will be registered by the router."
 - Button: "SIP PBX lines..."
- ISDN lines** section:
 - Text: "Here, you may configure the ISDN switching centers or phone systems for which the router itself is the end device. Outgoing calls may be made via Call Router on these lines."
 - Button: "ISDN lines..."
 - Text: "Here, you may assign an internal number to any MSN."
 - Button: "ISDN mapping..."

The screenshot shows the 'Dynamic SIP-Line - New Entry' dialog box with the following fields and options:

- Dynamic line name:** Text input field.
- SIP line name:** Dropdown menu with a 'Select' button.
- Priority:** Text input field with value '0'.
- Weight:** Text input field with value '0'.
- Algorithm:** Dropdown menu with 'Round-robin' selected.
- Max calls:** Text input field with value '0'.
- Buttons: 'OK' and 'Cancel'.

Dynamic line name

Enter the name for the dynamic line here. If the dynamic line consists of several physical lines, you can also use this dynamic line name for other table entries. This dynamic line name can later be used in the call routing table as the destination line.

SIP line name

Here you select one of the already configured physical SIP connections.

Priority

Here you specify the priority of the physical line for consideration when outgoing calls are distributed.

Weight

Here you specify the weighting of the physical line for consideration when outgoing calls are distributed.

Algorithm

The algorithm must be configured identically for all entries that belong to a dynamic line. The following algorithms can be used:

Weight

This algorithm controls the percentage of calls being distributed between different physical lines.

Round-Robin

With this algorithm, outgoing calls are distributed sequentially to the physical lines.

Priority

The physical line with the highest priority is fully utilized first, before the physical line with the next-lowest priority is used.

Max. calls

Here you enter how many simultaneous voice channels can be used on the physical SIP line. For no restriction on the number of voice channels, enter 0 here.

2.1.1 Additions to the Setup menu

Dynamic line

Configure the dynamic SIP lines here.

SNMP ID:

2.33.4.1.3

Console path:

Setup > Voice-Call-Manager > Line > SIP-Provider

Dynamic line name

Enter the name for the dynamic line here. If the dynamic line consists of several physical lines, you can also use this dynamic line name for other table entries. This dynamic line name can later be used in the call routing table as the destination line.

SNMP ID:

2.33.4.1.3.1

Console path:

Setup > Voice-Call-Manager > Lines > SIP-Provider > Dynamic-Line

Possible values:

Max. 32 characters from `[A-Z][0-9]@{|}~!$%&'()+-./:;<=>?[]^_.`

SIP line name

Here you specify one of the already configured physical SIP connections.

SNMP ID:

2.33.4.1.3.2

Console path:**Setup > Voice-Call-Manager > Lines > SIP-Provider > Dynamic-Line****Possible values:**Max. 32 characters from `[A-Z][0-9]@{|}~!$%&'()+-/,;=<=>?[]^_.`**Priority**

Here you specify the priority of the physical line for consideration when outgoing calls are distributed.

SNMP ID:

2.33.4.1.3.3

Console path:**Setup > Voice-Call-Manager > Lines > SIP-Provider > Dynamic-Line****Possible values:**Max. 3 characters from `[0-9]`**Weight**

Here you specify the weighting of the physical line for consideration when outgoing calls are distributed.

SNMP ID:

2.33.4.1.3.4

Console path:**Setup > Voice-Call-Manager > Lines > SIP-Provider > Dynamic-Line****Possible values:**Max. 3 characters from `[0-9]`**Algorithm**

The algorithm must be configured identically for all entries that belong to a dynamic line.

SNMP ID:

2.33.4.1.3.5

Console path:**Setup > Voice-Call-Manager > Lines > SIP-Provider > Dynamic-Line**

Possible values:**Weight**

This algorithm controls the percentage of calls being distributed between different physical lines.

Round-Robin

With this algorithm, outgoing calls are distributed sequentially to the physical lines.

Priority

The physical line with the highest priority is fully utilized first, before the physical line with the next-lowest priority is used.

Max. calls

Here you enter how many simultaneous voice channels can be used on the physical SIP line. For no restriction on the number of voice channels, enter 0 here.

SNMP ID:

2.33.4.1.3.6

Console path:

Setup > Voice-Call-Manager > Lines > SIP-Provider > Dynamic-Line

Possible values:

Max. 3 characters from [0-9]

2.2 Flex mode

From LCOS 10.34 the new Flex mode for SIP lines is supported. The configuration is found under **Voice call manager > Lines** by clicking the button **SIP lines**.

SIP lines

Here, you may configure lines for public SIP providers for which the router registers itself. Outgoing calls may be made via Call Router on these lines.

The SIP mapping table can be used to specify internal and external numbers for trunk and gateway SIP lines.

In the SIP PBX lines table you can define upstream SIP phone systems (PBX), for which all local users with a PBX-adequate domain will be registered by the router.

ISDN lines

Here, you may configure the ISDN switching centers or phone systems for which the router itself is the end device. Outgoing calls may be made via Call Router on these lines.

Here, you may assign an internal number to any MSN.

SIP lines - New Entry

General Security Advanced

Entry active

Mode:

Provider name:

Comment:

Provider data

SIP domain/realm:

Registrar (optional):

Port:

Switching at provider active

Login data

(Re-)Registration

SIP-ID/user:

Display name (optional):

Authentication name:

Password: Show

Call prefix:

Internal dest. number:

Mode

Flex


- > To the outside the line behaves like a commercially available SIP account with a single public number.

- > The number is registered at the service provider and registration is refreshed on a regular basis.
- > For outgoing calls, the calling-line number (sender) is not modified.
- > For incoming calls the dialed number (destination) is not modified.
- > The maximum number of connections at any one time is limited only by the available bandwidth.

2.2.1 Additions to the Setup menu

Mode

This selection specifies the operating mode of the SIP line.

 The "Service provider" can be a server in the Internet, an IP PBX, or a voice gateway. Please observe the notices about "SIP mapping".

SNMP ID:

2.33.4.1.1.17

Console path:

Setup > Voice-Call-Manager > Lines > SIP-Provider > Line

Possible values:

Provider

Externally, the line behaves like a typical SIP account with a single public number. The number is registered with the service provider, the registration is refreshed at regular intervals (if (re-)registration has been activated for this SIP provider line). For outgoing calls, the calling-line number is replaced (masked) by the registered number. Incoming calls are sent to the configured internal destination number. Only one connection can exist at a time.

Trunk

Externally, the line acts like an extended SIP account with a main external telephone number and multiple extension numbers. The SIP ID is registered as the main switchboard number with the service provider and the registration is refreshed at regular intervals (if (re-)registration has been activated for this SIP provider line). For outgoing calls, the switchboard number acts as a prefix placed in front of each calling number (sender; SIP: "From:"). For incoming calls, the prefix is removed from the destination number (SIP: "To:"). The remaining digits are used as the internal extension number. In case of error (prefix not found, destination equals prefix) the call is forwarded to the internal destination number as configured. The maximum number of connections at any one time is limited only by the available bandwidth.

Gateway

Externally the line behaves like a typical SIP account with a single public number, the SIP ID. The number (SIP ID) is registered with the service provider and the registration is refreshed at regular intervals (if (re-)registration has been activated for this SIP provider line). For outgoing calls, the calling-line number (sender) is replaced (masked) by the registered number (SIP ID in SIP: "From:") and sent in a separate field (SIP: "Contact:"). For incoming calls the dialed number (destination) is not modified. The maximum number of connections at any one time is limited only by the available bandwidth.

Link

Externally, the line behaves like a typical SIP account with a single public number (SIP ID). The number is registered with the service provider, the registration is refreshed at regular intervals (if (re-)registration has been activated for this SIP provider line). For outgoing calls, the calling-line number (sender; SIP:

"From:") is not modified. For incoming calls, the dialed number (destination; SIP: "To:") is not modified. The maximum number of connections at any one time is limited only by the available bandwidth.

Flex

- > To the outside the line behaves like a commercially available SIP account with a single public number.
- > The number is registered at the service provider and registration is refreshed on a regular basis.
- > For outgoing calls, the calling-line number (sender) is not modified.
- > For incoming calls the dialed number (destination) is not modified.
- > The maximum number of connections at any one time is limited only by the available bandwidth.

Default:

Provider