LocaPhone VoIP TK-System

Software Release Notes

Version 5.0.1

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LocaPhone VoIP TK-System Release 5.0

Release Notes - LocaPhone - Version 5.0.1
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Improved Functionality:

[PHONE-74]	[web-ui] use product name instead of "Gemeinschaft"
[PHONE-225]	[prov/yealink] feature key 'redial' uses server-based caller lists
[PHONE-273]	[astbuttond] default memory settings
[PHONE-320]	[prov/gigaset] N510IP phonebook usability
[PHONE-445]	[web-ui] layout/typo corrections to user interface
[PHONE-460]	[packaging] start t38modem at system startup
[PHONE-461]	[packaging] Remove unnecessary ntpdate package dependency
[PHONE-464]	[prov/yealink] new MAC vendor IDs for Yealink
[PHONE-479]	[prov/yealink] upgrade to firmware version 83

Bugs fixed:

[PHONE-279]	[web-ui] disabling phone types does not effect all GUI menus
[PHONE-316]	[packaging] Incomplete startup on fast machines
[PHONE-447]	[astbuttond] softkey profiles on snom do only work after astbuttond reset
[PHONE-457]	[livemon] user changes require astbuttond restart
[PHONE-462]	[core] backup script does not work anymore
[PHONE-476]	[dialplan] Call to IVR, forwarded to queue, forwarded somewhere else
[PHONE-480]	[prov/yealink] firmware for T4xS, T5x, CP920 missing
[PHONE-481]	[prov/yealink] Yealink models T5x missing in locaphone.php

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Release Notes - LocaPhone - Version 5.0.0
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New Functionality:

[PHONE-266] [phone-ui] limitation of call-list entries
[PHONE-305] [localization] language corrections in web-ui (German)
[PHONE-352] [prov/mitel] wrong/old firmware package for xenial
[PHONE-364] [web-ui] improved design for user interface
[PHONE-374] [scripting] improved CLI scripts for snom factory reset/reboot
[PHONE-378] [core] Retry SIP trunk registration on timeout
[PHONE-382] [applications] SnomRC 'auto refresh' checkbox
[PHONE-345] [prov/yealink] add support for Yealink CP920 conference phone
[PHONE-95] [dialplan] Call forward for hunt groups
[PHONE-190] [core][dialplan] Replace chan_sip by chan_pjsip in SIP gateways

Improved Functionality:

[PHONE-358] [web-ui] call log entries show month in English
[PHONE-441] [web-ui] update help page for softkeys
[PHONE-448] [phone-ui] improved default idle screen on snom D765/760
[PHONE-449] [web-ui] update help page for snom devices
[PHONE-264] [web-ui] Improve softkey configuration for snom D3xx with D3 extension
[PHONE-269] [prov/yealink] Better default microphones volumes
[PHONE-299] [dialplan] Improved transfer failed callback information
[PHONE-342] [phone-ui] Improved default screen colours on snom D765/760
[PHONE-360] [core] use local timezone in timestamp of CDR data
[PHONE-369] [prov/yealink] provisioning support for Yealink T58 firmware version 80
[PHONE-10] [prov/mitel] Mitel: change volume via web-ui
[PHONE-98] [provisioning] Use alarm call via phone menu
[PHONE-301] [livemon] sort extensions monitor by call time
[PHONE-302] [prov/yealink] support for Yealink T5x phones
[PHONE-309] [livemon] show all queues in detailed view of agent monitor
[PHONE-310] [prov/m800] Provision NTP server on OSN
[PHONE-115] [web_ui] delete manually added phones
[PHONE-117] [web_ui] check vendor MAC when adding phones into provisioning
[PHONE-191] Send and receive fax does not work between t38 and non t38
[PHONE-234] [prov/snom] remap 'help' key/menu item to help-info xml page
[PHONE-272] [M800] use VLAN 3992 configuration by default
[PHONE-278] [prov] Phone types not updated when provisioning devices

Bugs fixed:

[PHONE-330] [dialplan] call diversion does not work
[PHONE-442] [web-ui] key profiles PDF export broken
[PHONE-443] [web-ui] menu item 'IAX Gateways' cannot be hidden
[PHONE-444] [web-ui] call recordings play button does not work
[PHONE-446] [web-ui] key configuration 'locked' state indication missing
[PHONE-35] [web-ui] statistics pages do not work anymore
[PHONE-329] [dialplan] Unattended transfer of two outgoing external calls fails
[PHONE-373] [dialplan] Some voicemail messages are not being saved to database
[PHONE-375] [dialplan] Hotdesking by call leaves nobody user assigned to phone
[PHONE-379] [web-ui] Fix SIP trunk registration status
[PHONE-380] [web-ui] Characters are stripped from gateway SIP username/password
[PHONE-116] [web_ui] Removed M800 FXS ports from provisioning/phones
[PHONE-268] [prov/yealink] Fixed incorrect value range for microphone volume
[PHONE-280] [web-ui] Different minimum answer time when adding and editing IVRs
[PHONE-281] [core] Fixed incorrect REMOVEMEMBER events in queue log
[PHONE-287] [dialplan] Macro "macro-queue-answer-callback" does not exit correct
[PHONE-288] [core] Wrong path in musiconhold.conf
[PHONE-290] [provisioning] Set volume only for currently registered phone type
[PHONE-317] [packaging] logrotate of queue_log does not work
[PHONE-321] [fax] faxgetty stuck when no fax connection is established
[PHONE-325] [prov/snom] function 'group pickup' does not work
[PHONE-328] [core] backup does not include spool directory
[PHONE-331] [prov/snom] Hotdesking (XML) leaves nobody user assigned to phone
[PHONE-335] [dialplan] Incorrect number presentation when using parallel call
[PHONE-338] [livemon] Extensions monitor layout broken if too many users
[PHONE-339] [dialplan] Setting callforward to voicemail not working
[PHONE-341] [web-ui] Fixed incorrect presentation of recordings page
[PHONE-357] [dialplan] Fixed typo in gs_gateway_config_get.php
[PHONE-363] [prov/mitel] Mitel M685i key configuration does not work
[PHONE-368] Call diversion from queue to IVR and back does not work
[PHONE-370] [packaging] wrong firmware packages for Gigaset, Mitel devices
[PHONE-112] [web_ui] missing snom function key for function 'assisted transfer'
[PHONE-249] [dialplan] global calldrop timeout
[PHONE-275] [dialplan] Call deflection causes inbound caller-id to be replaced again
[PHONE-289] [core][dialplan] agents appear in queue "NONE"
[PHONE-291] [prov/mitel] 67xx/68xxi not available in volume settings menu
[PHONE-292] [prov/mitel] Aastra/Mitel minibrowser pages do not work anymore
[PHONE-300] [livemon] user changes require astbuttond restart
[PHONE-303] [prov/snom] Hold/Transfer softkeys missing on snom D305, D315, D345
[PHONE-313] [web-ui] Unable to add IVR
[PHONE-322] [dialplan] Group pickup does not work on queue calls
[PHONE-323] [core] Asterisk fails to load cdr_mysql module
[PHONE-324] [livemon] Unable to log off agent
[PHONE-326] [prov/snom] D375 shows wrong target extension if the call is transferred
[PHONE-71] [web_ui] Number of columns mismatch in external phonebook
[PHONE-245] [dialplan] CLIR does not work on SIP trunks
[PHONE-267] [dialplan] incorrect queue call logging when using custom dialplan
[PHONE-282] [dialplan] paging function causes syntax error in dialplan
[PHONE-285] [core] Voicemail not working
[PHONE-238] [core] incorrect PJSIP configuration parameter localnet
[PHONE-259] [web-ui] user rules cannot be added

LocaPhone VoIP TK-System Release 4.6

Release Notes - LocaPhone - Version 4.6.1
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Bugs fixed:

[PHONE-454] - [prov/mitel] Mitel M685i key configuration does not work
[PHONE-455] - [core][dialplan] agents appear in queue "NONE"
[PHONE-456] - [prov/mitel] 67xx/68xxi not available in volume settings menu
[PHONE-457] - [livemon] user changes require astbuttond restart
[PHONE-458] - [prov/snom] Hold/Transfer softkeys on snom D305, D315, D345 missing

Release Notes - LocaPhone - Version 4.6.0
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New Functionality:

[PHONE-178] JSON RPC interface for remote agent control
[PHONE-247] [web-ui] show configuration 'agent in queue(s)'

Improved Functionality:

[PHONE-167] [dialplan] queues with individual ringtones
[PHONE-171] [livemon] Extensions Monitor - set status DND, call diversion
[PHONE-198] [dialplan] Queue callforwards should resolve the queue name
[PHONE-219] [livemon] new features for agent/extension-monitor
[PHONE-221] [astbuttond/yealink] improve indication of DND/call infos on T4x
[PHONE-230] [prov/snom] Add exit button in snom dial log
[PHONE-233] [prov/yealink] support for Yealink T4xS phones

Bugs fixed:

[PHONE-120] [dialplan] TRANSFER_FAILED_CALLBACK does not work on no answer
[PHONE-158] [prov/snom] function keys not configured while phone is offline
[PHONE-161] [dialplan] Hangup causes will not be passed correctly
[PHONE-253] [prov/snom] multiple pause keys result in unpausing
[PHONE-254] [prov/gigaset] provisioning fails for Maxwell 3/basic
[PHONE-257] [dialplan] call forward does not work across cluster nodes

LocaPhone changes from version 4.5.1 to 4.6.0
==
* [livemon] PHONE-171 - Extensions Monitor - set status DND, call diversion
* [dialplan] PHONE-198: Replace the forwarded queue number with queue-name
* [prov/snom] PHONE-158 - function keys not configured while phone is offline
* [prov/gigaset] PHONE-254: Add missing changes for MX3B provisioning after the big refactor
* [dialplan] PHONE-120 - fixed callback of failed transfer in case of no answer
* [dialplan] PHONE-161 - fixed passing the SIP hangup cause from the callee to the caller
* [prov/snom] PHONE-230: Change order of softkeys and map the abort-key to cancel-button
* [web-ui] PHONE-247 - Added agents->queues overview page.
* [dialplan] PHONE-243: Replace the operations-range for identifier "CALL_CHECKDATE"
* [dialplan] PHONE-257 - call forward does not work across cluster nodes
* PHONE-178 - JSON RPC interface for remote agent control. Fixed typo.
* PHONE-178 Added JSON-RPC interface for LocaPhone
* [prov] PHONE-253 - Allow user to change pause reason without unpausing
* [core/queue] PHONE-167: Set an individual ring tone for calls from queues. This only works for snom phones.
* [prov/yealink] PHONE-233: Recognize and support Yealink T4xS devices
* PHONE-219 - Extended agent and extension monitor with additional features

astbuttond changes from version 4.5.0 to 4.5.1
==
* [livemon] PHONE-171 - Extensions Monitor - set status DND, call diversion
* [prov] PHONE-253 - Allow user to change pause reason without unpausing
* PHONE-233: recognize and support Yealink T4xS devices
* PHONE-219 - [livemon] new features for agent/extension-monitor

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Release Notes - LocaPhone - Version 4.5.1
==

Improved Functionality:

[PHONE-220]	[prov/yealink] increase phone display backlight idle timeout
[PHONE-223]	[prov/mitel] provide softkeys for 68xxi
[PHONE-231]	[prov/yealink] Do not delete call list without confirmation

Bugs fixed:

[PHONE-216]	[prov/yealink] Phones shouldn't use their 'internal' visual pickup info if astbuttond is in use
[PHONE-217]	[prov/gigaset] N720 umlaut breaks displayed names in phonebook
[PHONE-224]	[prov/mitel] firmware reporting broken
[PHONE-226]	[prov/yealink] The AstButtond must not configure pickup buttons for Yealink phones
[PHONE-228]	[prov/yealink] Toggling call forward sets timeout to 1 second

LocaPhone changes from version 4.5.0 to 4.5.1
==

* [prov/aastra] PHONE-223 - added default softkeys for DND, phonebook and voicemail on models 6869i and 6873i
* [prov/aastra] PHONE-224 - fixed firmware version reporting
* [prov/yealink] PHONE-228: copy timeout values to all forward cases
* [prov/gigaset] PHONE-75: fix API usage against commonized provisioning classes and functions
* [prov/yealink] PHONE-231: add confirmation screen before truncating call lists
* [prov/yealink] increase idle time until the phone display backlight is turned off
* [prov/yealink] Disable 'internal' visual pickup info if astbuttond is in use

astbuttond changes from version 4.5.0 to 4.5.1
===

PHONE-221	[astbuttond/yealink] improve indication of DND/call infos on T4x
PHONE-226	[prov/yealink] The AstButtond must not configure pickup buttons for Yealink phones
[prov/yealink]	The AstButtond does not configure pickup buttons for Yealink phones. Moreover the font size of the status text of model T48G is now larger and the font color is light gray.

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Release Notes - LocaPhone - Version 4.5.0
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Improved Functionality:

[PHONE-73]	[web_ui] Do not reveal email address on lost password
[PHONE-75]	[web_ui] [prov] Harmonization of call lists
[PHONE-94]	[prov/snom] function key to send DTMF tones
[PHONE-157]	[prov/aastra] add support for Mitel 6873i
[PHONE-197]	[prov/yealink] improved PBX integration (keys, system services)
[PHONE-204]	[core] Asterisk startup warning " No '=' (equal sign) in line 1 of /var/tmp/exec.1474360666291200.140021070284544"
[PHONE-207]	[prov] Removed deprecated provisioners

Bugs fixed:

[PHONE-181]	No update of screened display names and numbers for Yealink phones
[PHONE-186]	[core] gs-queue-callforward-activate, gs-queue-callforward-set are not working
[PHONE-193]	[prov/snom] keys on D7 extension of snom D765 are labeled with suffix "[inaktiv]"
[PHONE-195]	[core] Pickup of a queue call crashes asterisk

LocaPhone changes from version 4.4.0 to 4.5.0
==

* PHONE-197: Added missing ami event when activating changing the call forwarding.
* [prov/aastra] PHONE-157 - fixed phone identifier when adding new phones to the system
* [core/yealink] Don't do phonesync when astbuttond is in use
* [gui] Add additional Yealink softkeys, finish expansion module support
* [gui] Trigger Yealink-sync when missed calls get read and call forwards are updated
* [dialplan] Trigger Yealink sync on dialplan actions like missed calls
* [prov/yealink] VFS additions for XML application handling, add applications, add state resync URI
* [cli] Add CLI script to trigger Yealink phone sync (messages, LEDs) for an extension
* [core] Add system-wide Yealink support functions
* [prov/yealink] Helper class to aid in creating valid Yealink XML browser documents
* [prov/yealink] Add needed defines and methods to Yealink-GSFuncs, change include path to parent gsfuncs
* [prov/common] Add phoneuser logon/logoff handling functions (e.g. for performing hotdesking)
* [prov/aastra] PHONE-157 - added support for Mitel 6873i model. Refactored phone type detection because Mitel is using a new MAC vendor ID
* [webui] PHONE-73 - do not reveal email address when sending lost password email
* [prov/snom] PHONE-75: Change dial log lists to "all", "missed", "in" and "out" types by utilising the commonized diallog functions (same dial log as used by Gigaset), also provide a "truncate" softkey and optimize soft/hardkey button behaviour
* [prov/snom] "<" and ">" are fine in snom XML with current firmware versions
* [prov/all] Factor out common provisioning functionality to common/ and change provisioners to make use of it
* [prov/snom] PHONE-94: Add DTMF function key type
* [gui] Fix key configuration after 7f41e2ad94df889ff0c7acc42bd75f52b1e82eed
* [core] PHONE-204 - fixed Asterisk startup warning message and cleaned up code of Lua globals generator
* [cleanup] PHONE-207: Remove panasonic DECT code from everywhere
* [cleanup] PHONE-207: Drop unused twinkle and panasonic DECT provisioners
* [dialplan] PHONE-195 - prevent race condition that causes Asterisk to crash when picking queue calls
* [prov/yealink] PHONE-203: Add Yealink fwbin directory to tree
* [conf] PHONE-113: Bump default Yealink firmware versions
* [prov/yealink] PHONE-181: configure presentation of the caller ID to PAI/RPID/FROM
* [prov/snom] PHONE-193 - fixed key labels on D7 extension connected to snom D765
* [core] PHONE-186 - Added translation to new cf-rules to gs-queue-callforward-activate, gs-queue-callforward-set

astbuttond changes from version 4.4.0 to 4.5.0
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PHONE-157	[prov/aastra] add support for Mitel 6873i
PHONE-197	[prov/yealink] improved PBX integration (keys, system services). Added Yealink support to the AstButtond.

LocaPhone VoIP TK-System Release 4.4

Release Notes - LocaPhone - Version 4.4.0
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New Functionality:

[PHONE-41]	[prov/snom] snom M700/M325 DECT integration
[PHONE-137]	Support for Gigaset Maxwell Basic/3
[PHONE-143]	Paging function

Improved Functionality:

[PHONE-23]	LDAP installation script install-openldap.sh does not work with OpenLDAP 2.4.28-1.1ubuntu4.5
[PHONE-64]	[prov/yealink] Phone UI of nobody users is English
[PHONE-78]	[prov/snom] Hide undesired items from phone settings menu
[PHONE-86]	Snom MP access to caller lists
[PHONE-141]	Mail2Fax does not work
[PHONE-155]	[prov/snom] support for snom D3xx/D745 line of phones
[PHONE-156]	[documentation] update documentation for snom D3xx line of phones
[PHONE-170]	[prov/gigaset] CLIP update on unattended transfer
[PHONE-177]	[prov/snom] increase value of backlight_idle to 5
[PHONE-179]	[prov/snom] Packaging of current snom firmware 3xx/D3xx/7xx/D7xx

Bugs fixed:

[PHONE-70]	[prov/snom] Call list delete will always delete first entry
[PHONE-79]	[core] busy on busy is not working during queue calls
[PHONE-85]	[prov/snom] Unable to use XML applications on snom MP
[PHONE-92]	M800 mediagateway - no proper busy signaling
[PHONE-121]	[dialplan] Call completion prevents logging of missed calls
[PHONE-138]	[prov/mediant] No call progress on calls from BRI to FXS
[PHONE-163]	Gigaset Merkur devices: Putting calls on hold results in heavily distorted audio on the remote side
[PHONE-164]	[snom] disable 'browser cache' of XML mini browser
[PHONE-172]	No single quote characters in Live Monitor
[PHONE-175]	[prov/snom] provisioning settings snom Dxxx not working
[PHONE-176]	[prov/snom] Uncaught exception when invalid provisioning parameters are in use
[PHONE-182]	snom D375 reboots on user logoff/login
[PHONE-183]	[prov/polycom] provisioning for Polycom SoundStation IP 6000 not working
[PHONE-184]	[prov/snom] snom 300: OK key becomes ESC key in phonebook

LocaPhone changes from version 4.3.2 to 4.4.0
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* [prov/polycom-uc] add missing config file to actually make Polycom devices request <MAC>-phone.cfg
* [dialplan] PHONE-79 - fixed busy on busy during queue calls
* [prov/m800] PHONE-92: disable early-media for PSTN interfaces for proper call progress signalling (e.g. BUSY)
* [scripts] use "512@10.0.0.100" as recipient when doing check-sync by ip on snom DECT devices, allows for anonymous NOTIFY and makes hotdesking work
* [gui] Allow provisioning parameter profiles for Polycom UC-compatible devices
* [prov/polycom-uc] New provisioner for Polycom phones running UC firmware (> 4.0) with slightly enhanced functionality and a cleaner implementation without hundreds of kilobytes raw XML, fixes PHONE-183
* [prov/snomdect] snom M325 really identifies itself as M300, fix up strings accordingly, also fix provisioning minors related to the M300
* [prov/snomdect] fix warning if no subdevices were added yet
* PHONE-155 [prov/snom] support for snom D3xx/D745 line of phones
* [prov/snom] PHONE-78: add reduced main menu
* [prov/mediant800] PHONE-138: set up FXS users with early-media disabled
* [prov/snom] PHONE-86: map the full caller list menu onto the dial button on snom MP phones
* [gui] GUI additions/changes for snom Mxxx DECT device support
* [scripts] support snom Mxxx DECT devices in gs_prov_phone_checkcfg
* [prov/snomdect] autoprovisioning for snom M325/M700 single/multicell DECT devices
* [core/conf] configuration options for snom M325/M700 DECT device autoprovisioning
* [database] extend phones.mac_addr column to fit the <MAC>-<IPUI> combination used by the snomdect provisioner
* [conf] PHONE-179: fixup snom firmware versions according to updated packaging
* [gui] D305/D315/D345/D745: GUI additions to recognise and support new snom devices (e.g. softkey configuration), cosmetics
* [core/conf] D305/D315/D345/D745: Add firmware version options to LocaPhone configuration
* [prov/snom] D305/D315/D345/D745: Add bits to properly recognise and configure new phones, fix cosmetics
* PHONE-143 - Added paging function
* [prov/snom] PHONE-184: Don't remap softkeys in phonebook XML for requests from snom300 phones, fixes OK/CANCEL key mapping/usage
* [cleanup] Remove unused configfile and script cruft from asterisk config dir
* [cleanup] Clear cruft from htdocs/gui/mod
* [gui] Use long PHP open tags
* [compat] lua5.2: fix up naming conflict with the goto application, fix escape sequences in scripts and lua globals generator
* [compat] lua5.2: replace table.getn with the length operator (table.getn was already deprecated in lua5.1)
* PHONE-182: Recognize snom-Dxxx phonetype strings in gs_prov_phone_checkcfg, fixes reboot on user login/logout

LocaPhone changes from version 4.3.2 to 4.4.0
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* [prov/gigaset] improve filter parameter evaluation in
  gigaset_gs_vmbox_getmsglist()
* [core/scripts] gs_prov_phone_checkcfg: Remove two invalid headers from
  Gigaset Corded IP sync, fixes reprovisioning of MX3B devices
* [core/scripts] Do Gigaset XMLif data sync also for gigaset-maxwell* phone types
* [gui] Recognise and support Gigaset Maxwell Basic/3 in GUI parts
* [prov/gigaset] Add Maxwell Basic/3 support to the Gigaset provisioner
* [core/conf] Add configuration options for Gigaset Maxwell 3/Basic devices
* [dialplan] PHONE-121 - Calls that are forwarded to a call completion
  extension will be logged as missed for the originally called user.
* PHONE-141 - Added mail2fax support for easy use with LocaPhone.
* [prov/snom] PHONE-176: catch DOMDocument exceptions to prevent
  provisioning parameters with illegal xml characters breaking configuration creation
* PHONE-175 - Provisioning parameters will now be applied to Snom Desk
  Phones (D-series).
* [prov/gigaset] PHONE-170 - allow CLIP update on unattended transfer on N720
* [prov/gigaset] PHONE-163 - merkur: explicitly configure sendonly as hold method
* [core] PHONE-23 - updated OpenLDAP phonebook integration to use existing
  LDAP structure
* [prov/snom] Improved backlight settings for snom phone models with color display

astbuttond changes from version 4.3.2 to 4.4.0
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PHONE-155	Added support for new snom phones and extension modules (D3xx/D745/D3)
PHONE-172	Fixed displaying single quote characters in the Live Monitor.

LocaPhone VoIP TK-System Release 4.3

LocaPhone changes from version 4.3.1 to 4.3.2
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* PHONE-147 - Fixed various issues that prevented Aastra/Mitel deskphones from being configure in a proper way.
* [dialplan] PHONE-122 - allow calling a queue without a timeout if no call-forward timeout has been set
* [prov/aastra] PHONE-151 - added support for Mitel 6863i phone
* [prov/aastra] PHONE-124 - Added softkeys for transfer and conference because some phone models do not have a hard key for these functions
* [prov/aastra] PHONE-107 - added support for Aastra 6735i and Aastra 6737i models
* PHONE-146 - The resolver can now determine the correct dial timeout for queues, even if call forward condition "all busy" is set.
* [webui] PHONE-108 - added missing Mitel 68xxi phone models to provisioning/phones page
* [core] PHONE-56 - added missing group nobody_users to database
* PHONE-145: [gui] key-layout: fix PDF creation for snom D7* phones
* PHONE-129 - Fixed evaluation of two digit extensions that are starting with the fax prefix.
* PHONE-123 - Reimplemented playing the voicemail announcement only.
* PHONE-133 - Fixed displaying the queue number when calling a queue from internal.
* [locaphone/luadialplan] PHONE-97 - For nobody users no dial log will be written
* [prov/snom] PHONE-127/PHONE-130/PHONE-131: Add branding for snom-D375, fix branding for snom-D765
* [prov/snom] PHONE-62: Remove unneeded emergency numbers from keybord_lock_emergency
* [prov/m800] PHONE-128: Bump default firmware version to 6.60A.304.001
* [dialplan] PHONE-52/PHONE-53: Add timestamp to recording filenames, output timestamp in GUI
* [web_ui] PHONE-109 Renamed Aastra/Mitel 68xx to only Mitel 68xxi

astbuttond changes from version 4.3.0 to 4.3.2
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* PHONE-111 - Added support for Mitel/Aastra 68xxi and 67xxi. Moreover added extension M680i and M685i.
* PHONE-132 - After logging off and on on a phone and after restarting the astbuttond the state of the queue buttons will be restored.
* PHONE-139 - Added new thread that updates the missed calls of each user (if needed) instead of reloading the whole service. So the daemon will not terminate unexpectedly.
* PHONE-60 - Replaced character `'` by `"` in the monitoring application because the ape service can not quote this character.

LocaPhone changes from version 4.3.0 to 4.3.1
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* [gui] PHONE-101 - Fixed CDRs cannot be filtered by call outcome
* [locaphone/luadialplan] PHONE-105 - Fixed Call forwarding rules with time and non-time conditions
* [locaphone/luadialplan] PHONE-51 - Fixed dial status for unallocated incomming and outgoing calls.
* [locaphone/luadialplan] PHONE-91 - Fixed channel state for busy calls in cdrs.
* [dialplan] PHONE-84 / PHONE-90 - Fix: Call forward of empty queue does not work
* [locaphone/provisioning] PHONE-89 - Fixed storing provisioning parameters for snom D375 and snom D765
* [locaphone/gui] PHONE-83 Fixed PDF key labels for snom D765
* PHONE-87 - Fixed typo in extensions.lua
* [prov/gigaset] PHONE-28: Recognise C430(A) GO devices
* [core/db] PHONE-81: Add missing value for gates.autorecord, add proper values to gate_params for the default SIP gateway
* [gui] Add tabindex properties to all input fields in inc_keyprof

LocaPhone changes from version 4.2.3 to 4.3.0
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* [locaphone/luadialplan] PHONE-68 - do not replace callerid name on inbound external calls with the callerid number if no contact can be found in the phonebook or reverse search has been disabled
* [doc/db] Name UPGRADE-DB properly for 4.3 release
* [gui] PHONE-53: Add webinterface for managing (listen, download, delete) recorded calls
* [gui] Create VM spool userdir if non-existent when uploading VM announcements
* [locaphone/alarmsystem] PHONE-65 - Define custom ring tone for alarm calls
* [core] PHONE-52: Provide option to record (MixMonitor) all calls going from/to a SIP gateway
* [locaphone/luadialplan] PHONE-55 - Fix: Callforwarding time does not reset after transferring picked up external call back to its origin called extension
* [locaphone/luadialplan] PHONE-59 - Fixed call forward/call timeout issue
* [locaphone/luadialplan] PHONE-66 - fixed usage of second failover gateway group for outbound calls
* [locaphone/luadialplan] PHONE-63 - fixed blind transfer failed function

* [locaphone/dialplan] PHONE-48 - Fixed call recording in huntgroups
* [prov/yealink] Fix notify action id in prov_phone_checkcfg_by_ext, don't force keepipadr option when calling prov_update_ip
* [doc/ldap] PHONE-24: auto-determine location of LDAP related files
* [prov/snom] PHONE-58: provision backlight_idle
* [gui] Add subpage for management of the voicemail greeting/announcement
* [locaphone/dialplan] PHONE-61 - Remove sip header "Privacy" before setting it again.
* [prov/snom] Disabled short SIP headers
* [prov/aastra] Updated language pack to version 4.1.0.128
* [locaphone/provisioning] Added snom D375 and snom D765 support
* [locaphone/core] You can control if the astbuttond will send display messages to a group of phones by adding the permission `push_message` to the related group of users. You can also control if the astbuttond sends queue information to the displays of the phones by using the same permission in a group of queues (see svn commit 1565 and 1579).
* [locaphone/luadialplan] Removed different voicemail announcements for internal/external calls. The unavailable message will now be used as the voicemail greeting if it has been recorded via the voicemail voice menu
* [prov/yealink] Autoprovisioning for Yealink SIP-T4x IP phones
* [prov/yealink] GUI additions for Yealink IP phone support
* [prov/yealink] Script additions and SIP NOTIFY event for Yealink IP phone support
* [prov/yealink] Configuration parameters for Yealink IP phone provisioning
* [core/gui] Provisioning/Phones: Editable user-assignment list, add form to manually add new phones (MAC address and type, helpful for PROV_AUTO_ADD_PHONE=false), reorder columns

locaphone4.0 (4.3.0-precise1) unstable; urgency=low

astbuttond changes from version 4.2.x (r1602) to 4.3.0
===

* INFRA-39 - Changed version info file name from .svn-revision to .get-revision
* Adjust the project to use git and Netbeans 8.0.2
* Added snom D765 support.
* Added support for snom D375 with "Expansion Module V2.0"
* Push the current inbound profile name by permission "display_profile_in"
* Fixed debug console command "peer show keys"
* Added assisted transfer function.
* You can switch on/off display messages for peers by using the permission system of locaphone (permission `push_message`). Moreover you can also switch on/of queue status informations with the same permission.

LocaPhone VoIP TK-System Release 4.2

locaphone4.0 (4.2.3-precise1) unstable; urgency=low

* Upstream version bump to 4.2.3 (bugfixes)

* LocaNet branch git/svn changes/additions since 4.2.2:

  - [doc] Add sample remote log config file for rsyslogd
  - [prov/mediant800] Configurable syslogging options
  - [prov/aastra] Added support for Aastra/Mitel 6800 family of phones
  - [prov] PHONE-34: Don't force-reboot in cluster environments
  - [core] Add missing title and prefix parameters to the gs_huntgroup_add() call in to-v4-migrate-huntgroups.php, fixes upgrading after a1e6ce6af7556747fae89f2af31e497e766f54b2
  - [locaphone/luadialplan] PHONE-47 - Fixed distribution of system recordings to cluster nodes
  - [locaphone/luadialplan] PHONE-45 - Fixed considering the cluster location of a queue when trying to log in
  - [locaphone/luadialplan] PHONE-46 - Fixed triggering a call to a phone from a foreign cluster node
  - [locaphone/luadialplan] PHONE-37 - Fixed call pickup across cluster nodes.
  - [locaphone/core] PHONE-42/43 Fixed unattended transfer from huntgroup users to huntgroups.
  - [prov/snom] Fix snom300 dial screen softkeys after 553cc3847ad5e03a8e7fab585d1e7cb53d1d5a91 for FW>=8.7.4.7
  - [locaphone/provisioning] Fixed trimming the users idle screen name.
  - [locaphone/luadialplan] PHONE-33 - Fixed references to the extension resolver in `from-node` context.
  - [locaphone/luadialplan] Fixed callforward condition `internal`->`no answer` for users
  - [locaphone/luadialplan] Fixed exception if quickdial number does not exist

locaphone4.0 (4.2.2-precise1) unstable; urgency=low

* Upstream version bump to 4.2.2 (bugfixes)

* LocaNet branch git/svn changes/additions since 4.2.1:

  - [prov/gigaset] N720: Only announce FW binaries if provided versions are higher than the basestation supplied version
  - [core] PHONE-25 - fixed hold time announcement in queues
  - [prov/gigaset] N720: Disable G.722 (wideband) usage
  - [prov/m800] Add required config options for FW297.003
  - [locaphone/luadialplan] Bug#0000811 - Fixed forwarding of calls with external origin to a parallel call target that contains an external destination. Since call completion never work reliable with parallel call targets call completion has been disabled.
  - [conf] Set snom FW versions to 8.7.5.17 (360: 8.7.3.25.9)
  - [prov/scripts] Add script to trigger snom3xx factory reset for phones tracked in the phones database table
  - [prov/snom] quirk: work around fw8.7.5.17 key labeling uglyness
  - [locaphone/dialplan] Fixed inbound routing: Disabled entries with inbound routing profile `0` will be ignored.
  - [locaphone/fax-integration] Read database credentials from configuration
  - [prov/gigaset] N720: Recognize user agent strings with appended MAC address as well
  - [locaphone/luadialplan] Write a missed call to the dial-log when a call is forwarded to the voicemail box.
  - [locaphone/luadialplan]Fixed potential null pointer error in the lua dialplan.
  - [prov/gigaset] Fix i18n setup for cordless devices
  - [locaphone/core] Fixed potential syntax error in configuration file
  - [locaphone/provisioning] FIX: Trimmed the idle screen user name to avoid scrolling of the name (snom phones only).
  - [locaphone/provisioning] Fixed software version detection from user agent string for snom phones (Did not work with firmware 7.3.30).
  - [prov/snom] Provision callwaiting per-identity for FW8.7.5+, provision always_show_active_call for proper callwaiting display
  - [prov/gigaset] N720: Disable conferencing function
  - [prov/gigaset] Merkur/Einstein: Init i18n so localization will be applied

locaphone4.0 (4.2.1-precise1) unstable; urgency=low

* Upstream version bump to 4.2.1 (bugfixes)

* LocaNet branch git/svn changes/additions since 4.2.0:

  - [core/db] Fix missing value in asterisk.sql for the ast_sipfriends table
  - [locaphone/luadialplan] Fix: Calls that are initialised by the phone system, e.g. a wakeup call, will not fail with a dialplan error.
  - [locaphone/luadialplan] You may forward a call to a quick dial number.
  - [locaphone/luadialplan] Fix: You may now transfer external calls unattended to an external target.

locaphone4.0 (4.2.0-precise1) unstable; urgency=low

* Upstream version bump to 4.2.0 (feature release)

* LocaNet branch git/svn changes/additions since 4.1.3:

- [prov/mediant800] Change CED transfer mode and Fax signaling method for better T.38 compatibility
- [core/cfg] Bump snom default FW version to 8.7.5.13
- [prov/snom] Fix provisioning of the retrieve button on FW8.7.5+
- [prov/snom] settings: Provision branding-related configuration items
- [prov/snom] branding: Mechanism and functions to handle (safe) delivery of branding-related files
- [prov/snom] branding: Replacement idle screens and splash/background graphics for LocaPhone branding
- [prov/snom] snom-common: Function to reply a HTTP 404 while canceling script execution, add define for the snom provisioning fs-root
- [prov/snom] cSnomCFG: Provide functionality to populate the uploads-category in XML config files
- [prov/snom] sw-update: Remove unneccessary user request var
- [prov/snom] Make use of refactored UA/MAC parser functionality in settings.php and sw-update.php, further cleanups
- [prov/snom] Move UA parser and MAC validator to snom-common, add wrapper for both functions
- [prov/snom] Removed functionless directory pool from dial screens and from call screen. Configuration menu is now available on "down key" across all phone models.
- [locaphone/provisioning] DND and call list buttons will also be controlled by the Astbuttond
- [prov/aastra] Configuration variable to append additional hosts to "xml application post list"
- [gui/keylabel] Support label printout for snom D725 phones
- [prov/gigaset] N720: Provision "BS_IP_Data.ucBAllowNotifyFromAnonymous" (neccessary since latest firmware versions, fixes non-working config refetch request via SIP NOTIFY)
- [prov/snom] Extension module (snom D7) support for snom D715
- [prov/snom] Support for snom D715, snom D725, snom 870, snom MP, snom PA1
- [gui] Expand short language names sent by newer browsers (esp. firefox)
- [prov/snom] Add extension module (Snom D7) support for snom720/760
- [core] Make SIP video support a per-user configurable option
- [prov/snom] Write fkey labels into label attributes of fkey XML elements for FW8.2+
- [prov/snom] De-duplicate HTTP header set up and nonchunked transfer enforcing
- [prov/snom] cleanup, unclutter, cosmetics, commentary, readability for settings and sw-update
- [prov/snom] Make codec order identical for all firmware versions
- [prov/snom] Drop old (and ugly) way of generating the settings file, utilize cSnomCFG instead
- [prov/snom] Get rid of _snomcnfXmlEsc and $prov_url_snom, use function and define from snom-common
- [prov/snom] Provide function for (snom-specific) XML escaping in snom-common
- [prov/snom] Class for handling the snom configuration file
- [prov/snom] Language files: Drop language version list, rather require a filled lang-<version> dir for (provisioned) firmware versions
- [prov/snom] config: Bump default FW version for all phones to FW 8.7.3.25.9
- [prov/polycom] Set 24h clock option also for phones running FW 2.1.2 (or earlier)
- [prov/gigaset] Maxwell: Change UA regex to recognize any DVF revision with also alphanumeric in version parts
- [prov/snom] FW-upgrade: Drop 6to7 upgrade, cleanup sw-update.php
- [prov/snom] Refactor firmware version string handling (now supports version strings with an arbitrary number of parts and letters)
- [prov/snom] Introduce snom-common.inc.php with various functions and defines usable in the snom provisioning
- [locaphone/core] Reimplemented alarm system.
- [locaphone/core] Added static queue members
- [prov/gigaset] Maxwell: Fix undefined constant notice
- [prov/gigaset] Maxwell: Properly support firmware versions consisting of four parts
- [prov/gigaset] Maxwell 10 support: Enable H.264 support in Asterisk's SIP configuration
- [prov/gigaset] Maxwell 10 support: Configuration options
- [prov/gigaset] Maxwell 10 support: Required changes to core components (scripts, GUI etc.)
- [prov/gigaset] Support and provision Maxwell 10 devices incl. management of device firmware (minimum required firmware version is 1.0.28)
- [locaphone/core] Added title and callerid-prefix for huntgroups.
- [locaphone/phone-ui] Added wakeup call xml setup page for snom, aastra and gigaset phones.
- [locaphone/dialplan] Added inbound routing profiles. The profiles are selectable by a service number. Added permission `set_routing_profile` and included the permission into the web ui.
- [core/gui] Allow up to 22 ringtones to be selected, add Gigaset note to ringtone selection module
- [dialplan/exttype-user] Refactor setRinger(): Read phonetype separately for generic ringer handling, apply DRP for Gigaset phones, cosmetics
- [prov/gigaset] Provision ringtone selection from user settings
- [prov/gigaset] Add support for C430 IP DECT basestations, add identifiers for A-variants (basestations w/internal answering machines)
- [prov/polycom] Add gitignore, setting: Configure backlight
- [prov/polycom] settings: Properly provision language and date/time settings
- [prov/polycom] features: Add missing includes (fixes broken CID selection)
- [prov/polycom] settings: Provision 'voicemail' (instead of 'mailbox')

as mailbox ext (fixes calling voicemail using the phone's 'messages' button)
- [prov/polycom] directory: Remove hardcoded ua override (probably used for debugging purposes), utilise (Gigaset) API to retrieve and populate the non-microbrowser-phonebook
- [prov/polycom] extnumber-display: Fix doctype var usage, cleanups and cosmetics
- [prov/polycom] callforwards: Fix CFW settings subpage (duplicate/illegal output removed)
- [prov/polycom] dnd: Use API for DND control, simplify/cleanup code, cosmetics
- [prov/polycom] phonebook: Use (Gigaset) API, cleanup and simplify code
- [prov/polycom] diallog: Make use of (Gigaset) API functions, improve and tidy up (and get rid of almost all SQL) code
- [prov/polycom] Cosmetics and cleanups
- [prov/polycom] htaccess: Remove RewriteBase and absolute paths
- [prov/gigaset] Function to convert diallog type strings to their respective flags used in diallog filter

LocaPhone VoIP TK-System Release 4.1

locaphone4.0 (4.1.3-precise1) unstable; urgency=low

* Upstream version bump to 4.1.3

* LocaNet branch git/svn changes/additions since 4.1.2:

  - [locaphone/luadialplan] Removed queue callback macro because it may cause Asterisk to crash
  - [locaphone/core] Fixed Bug 807: fixed backup and made it log to syslog
  - [locaphone/core] Reintegrated and improved fax integration with lua dialplan
  - [locaphone/dialplan] Fixed call forwarding and parallel/huntgroup calls. Never ever forward calls with callfile origin.
  - [galilei/dialplan] The channel variable `huntgroup-prefix` will not be set to a nil value when calling a parallel call extension.
  - [locaphone/core] Added userevent for notifying the astbuttond that the users name has been changed.
  - [prov/gigaset] Add i18n for strings used in xmlInterface/gc
  - [locaphone/luadialplan] Removed talker detection from conference rooms
  - [locaphone/luadialplan] Fixed setting the timeout for queues in the lua dialplan forwarding rules.
  - [locaphone/core] Added sending new userevent when adding a new user.

locaphone4.0 (4.1.2-precise1) unstable; urgency=low

* Upstream version bump to 4.1.2

* LocaNet branch git/svn changes/additions since 4.1.1:

  - [locaphone/callcenter] You are now able to pause an agent directly after logging in.
  - [prov/mediant800] Set up ignore-bri-los-alarm parameter depending on configured BRI port types
  - [locaphone/luadialplan] Fixed hangup cause for external calls
  - [locaphone/monitoring] The monitoring services (ape) will work without wildcard dns entries.

locaphone4.0 (4.1.1-precise1) unstable; urgency=low

* Upstream version bump to 4.1.1

* LocaNet branch git/svn changes/additions since 4.1.0:

  - [prov/mediant800] Enable early-media for SIP, enable ringback-tone playback for cases where the PSTN doesn't provide proper signalling
  - [prov/m800] Provision incoming call behaviour bits (USER_SCREEN_INDICATOR bit added to default value)
  - [prov/m800] Refactor generation of ip2tel mapping tables for proper callerid signalling especially when handling redirects on CLIP no screening lines
  - [prov/m800] Improved T.38 and modem relay settings
  - [locaphone/core] Removed infinite loop from gs-lua-globals-gen
  - [locaphone/luadialplan] Fixed boss-secretary function
  - [locaphone/luadialplan] Fixed Bug#774: You can now transfer a picked call to the original extension.
  - [locaphone/luadialplan] Fixed Bug 775: The system now also uses the caller ids that are defined in the gateway groups for different extensions.
  - [locaphone/luadialplan] Added delimiter `-` after the extension when calling the pickupchan-command to make sure that the correct channel will be picked.
  - [locaphone/luadialplan] Use an alternative move command to be able to move a file between different filesystems.
  - [locaphone/luadialplan] fixed call completion bugs #613, #630, #696
  - [mediant800] Fix reload of Asterisk configuration data (and a blank WebGUI page) when changing the Mediant800 settings
  - [locaphone/voicemail] Removed product name from email. Fixed From header if GS_EMAIL_NAME contains spaces.
  - [locaphone/luadialplan] fix for callfile (spool call) and parallel calls
  - [locaphone/dialplan] callfile possible on dnd
  - [prov/gigaset] Make firmware update method (TFTP push or HTTP download) configurable, implement bits in master.bin handling
  - [prov/gigaset] Print TFTP push transfer progress to the PBX log
  - [prov/gigaset] Add timeout and safety measures into TFTP push mechanism for proper cleanup and handling upon failure or timeout

locaphone4.0 (4.1.0-precise1) unstable; urgency=low

* Upstream version bump to 4.1.0

* LocaNet branch git/svn changes/additions since 4.0.0:

- [conf] Bump Gigaset firmware versions
- [Docs] Add instructions to start the Mediant 800 MGW autoconfig safely
- [GUI] Simple WebUI module at Routing/Mediant800 for browser-based M800 configuration, GUI enhancements for M800 devices
- [prov/m800] Database definition for m800-specific configuration items
- [prov/m800] Configuration variables
- [prov/m800] Functions and (global/system) commands
- [prov/m800] Call progress tone binary files
- [prov/m800] Autoprovisioning for Audiocodes Mediant 800 MSBGs
- [prov/gigaset] Recognize/support N720 devices in gs_prov_phone_checkcfg
- [prov/gigaset] Add Gigaset N720 support to the WebGUI
- [prov/gigaset] N720 configuration variables
- [prov/gigaset] Add support for Gigaset N720-DM/IP multicell DECT systems
- [prov/gigaset] Merkur: Show user-readable call-causes on handsets
- Update gsdb.lua
- [locaphone/luadialplan] Fixed sql error in permissions.lua when calling a nobody user
- [locaphone/luadialplan] Fixed Bug#751: We are now signaling the proper sip status on hangup.
- [locaphone/luadialplan] Bugfix: workaround for missing URL-deconding of hash sign in dialplan. Required for dialog-based call forward (*2#).
- [locaphone/luadialplan] Fix: Inbound routing table will also be used if there is no search/replace string defined for the called number in the gateway group configuration.
- [luadialplan] removed leading slash from sounds directory
- [locaphone/luadialplan] added utf8 connection string to database connect and example config for /etc/mysql/conf.d to fix utf8-problems with snom-phones
- [prov/gigaset] Updates and support regarding Merkur (DECT) support in scripts and WebGUI files
- [prov/gigaset] Database structure updates for increased length requirements of the MAC address field to handle Merkur (and other) handset slots
- [prov/gigaset] Webserver configuration updates/enhancements for Merkur (DECT) support
- [prov/gigaset] Configuration option changes/updates for Merkur (DECT) support
- [prov/gigaset] Support for Merkur (cordless/DECT) devices
- [gigaset/prov] Support/recognise another MAC vendor ID range
- [gigaset/xmlif] Mailbox: Implement message playback handling and callback file generation
- [gigaset/xmlif] Mailbox: Announce multiple user mailboxes if applicable
- [luadialplan] Add vmplayer.vmplay* context/extension for direct vm message playback, provide mailbox sync method in core.user
- [locaphone/provisioning] Fix: Added provisioning option for snom phones so the phone will prefer the provisioning url from the dhcp server. Logging in/out a user on a phone will work more certain if the phone can not reach the snom redirection service.
- [gigaset/xmlif] Enforce entry amount limit on unfiltered CallList read request (should fix crashing on DE310/410 with too huge unread missed calls history)
- [locaphone/core] Fixed path in /etc/locaphone/asterisk/README.txt
- [locaphone/core] Bugfix: rotate logfiles in /var/log/locaphone instead of legacy directory
- [locaphone/db] fixed wrong alter table, mantis bug 623
- [locaphone/luadialplan] Calls to huntgroups with external targets can now fail (busy/no answer/canceled) without causing a database error.
- [locaphone/luadialplan] fix for mantis-bug #615 (call forking and diallog)
- [locaphone/luadialplan] calldrop fix for mantis-bug #614
- [locaphone/luadialplan] fix for parallelcall mantis-bug #616
- [locaphone/luadialplan] fix for situation where gwgroup is empty
- [locaphone/luadialplan] Fix: The phone system can now determine the proper source of a call and playback the right voice mail announcement.
- [locaphone/database] New feature: Added primary key to table `dial_log`.
- [gigaset/xmlif] CallLists: Use entrynum of original result instead of afterwards-guessed id for XML entry ids in call lists, operate on full diallog list instead of filtered when deleting entries (fixes single-entry-delete always deleting in missed list)
- [gigaset/gsfuncs] Track entry index upon diallog retrieval
- [shared/scripts] Update/Fix includes from /opt/gemeinschaft to /opt/locaphone, use GS_DIR

LocaPhone VoIP TK-System Release 4.0

locaphone4.0 (4.0.0-precise1) unstable; urgency=low

* Upstream version bump to 4.0.0

* LocaNet branch git/svn changes/additions since 3.90.4:

  - [gigaset/prov] Support/recognise another MAC vendor ID range
  - [locaphone/provisioning] Fix: Added provisioning option for snom phones so the phone will prefer the provisioning url from the dhcp server. Logging in/out a user on a phone will work more certain if the phone can not reach the snom redirection service.
  - [gigaset/xmlif] Enforce entry amount limit on unfiltered CallList read request (should fix crashing on DE310/410 with too huge unread missed calls history)
  - [locaphone/core] Fixed path in /etc/locaphone/asterisk/README.txt
  - [locaphone/core] Bugfix: rotate logfiles in /var/log/locaphone instead of legacy directory

locaphone4.0 (3.90.4-precise1) unstable; urgency=low

* Upstream version bump to 3.90.4 (4.0beta4)

* LocaNet branch git/svn changes/additions since 3.90.3:

  - [locaphone/luadialplan] Calls to huntgroups with external targets can now fail (busy/no answer/canceled) without causing a database error.
  - [locaphone/db] fixed wrong alter table, mantis bug 623
  - [locaphone/luadialplan] fix for mantis-bug #615 (call forking and diallog)
  - [locaphone/luadialplan] calldrop fix for mantis-bug #614
  - [locaphone/luadialplan] fix for parallelcall mantis-bug #616
  - [locaphone/luadialplan] fix for situation where gwgroup is empty
  - [locaphone/luadialplan] Fix: The phone system can now determine the proper source of a call and playback the right voice mail announcement.
  - [gigaset/xmlif] CallLists: Use entrynum of original result instead of afterwards-guessed id for XML entry ids in call lists, operate on full diallog list instead of filtered when deleting entries (fixes single-entry-delete always deleting in missed list)
  - [gigaset/gsfuncs] Track entry index upon diallog retrieval
  - [shared/scripts] Update/Fix includes from /opt/gemeinschaft to /opt/locaphone, use GS_DIR

locaphone4.0 (3.90.3-precise1) unstable; urgency=low

* Upstream version bump to 3.90.3 (4.0beta3)

* LocaNet branch git/svn changes/additions since 3.90.2:

  - [locaphone/callforwards] Fixed bug#577: Call forwardings to the voicemail box (on "no answer" and on "busy") now work together with call forking. Fixed determining the default dial timeout for users/huntgroups.
  - [locaphone/luadialplan] fixed conditions in datetime callforward module
  - [locaphone/luadialplan] removed callforward condition
  - [locaphone/luadialplan] fix for #601, wrong hangupcause when offline and forward target is busy
  - [locaphone/gateways] Fixed bug#588. After removing a sip gateway the asterisk-service will reload to remove the gateway from its runtime configuration.
  - [locaphone/agents] Added new function to login and logout agents. This function can be used by php-scripts, and also fixes bug#617 (assigning multiple agents to a single user). Integrated new function into agent login/logout-script for snom- and aastra-phones.
  - [locaphone/backengine] Fix: When deleting users and queues all references in the database are removed correctly. This also fixes bug#569.
  - [locaphone/luadialplan] Added missing include in huntgroup.lua.
  - [locaphone/luadialplan] Fixed several bugs in call completion:
    - missing default value for parameter CC_TIMEOUT
    - call completion not offered if called party is on DND
    - fixed dialplan loop in modul exttype-callcompletion
    - cc-worker calculates the completion timeout correctly
  - [locaphone/luadialplan] Fixed bug#611: Unanswered queue calls that were forwarded to another target (e.g. a voicemail box) have not been logged as missed queue calls.
  - [locaphone/luadialplan] Bug #608 - Fixed dialplan so that calls to an extension with call forking activated will not be written to the call log multiple times.
  - [luadialplan] Move diallog-sync to function, trigger also for huntgroup and queue ext types
  - [locaphone/provisioning] added panasonic dect provisioning
  - [webui/gigaset-prov] Fix: Don't depend on astbuttond when data synchronisation is neccessary
  - [locaphone/luadialplan] Added missing group pickup extension that is needed by aastra phones.
  - [locaphone/provisioning] Fixed rewrite rules in `/opt/locaphone/htdocs/prov/polycom/.htaccess`. Renamed `gemeinschaft` to `locaphone`.
  - [luadialplan] Group hints: Group hint generator, include in extensions.lua
  - [locaphone/gui] fixed bug in pb-csv-export when impersonating as other user
  - [locaphone/luadialplan] removed hardcoded path and replaced with dynamic code
  - [luadialplan] Rename dbfuncs:createHints() to createUserHints(), refactor to pass and act/return possibly initialised table

- [prov/gigaset] Gigaset integration into LocaPhone WebGUI (provisioning options, key config etc.)
- [prov/gigaset] LUA-dialplan additions, Asterisk configuration changes
- [prov/gigaset] Generic and Gigaset-specific additions to LocaPhone scripts
- [prov/gigaset] Gigaset-specific LocaPhone configuration options and vars
- [prov/gigaset] Apache configuration for Gigaset PBX-TE XML interface
- [prov/gigaset] Scripts and local configuration to provide VFS for base configuration and PBX-TE XML interface
- [locaphone/luadialplan] Fixed Bug#597: When picking a queue call from a user the originally called user is busy until the picked call ended.
- [locaphone/setup] Added script to copy files of installed LocaPhone software and configuration files to DRBD volume.
- [locaphone/core] directory name substitution in /etc/locaphone/locaphone.php
- [webgui/keyprov] Remove remainder of Samsung provisioning
- [locaphone/core] Changed default SIP callerid and realm

locaphone4.0 (3.90.2-precise1) unstable; urgency=low

* Package-rename from gemeinschaft4.0 to locaphone4.0

gemeinschaft4.0 (3.90.2-precise1) unstable; urgency=low

* [locaphone/agentmonitor] Added info that the agent is in wrapup time.
* [locaphone/agentmonitor] Added showing the agents penalty
* [locaphone/core] transition from product Gemeinschaft to LocaPhone
* [locaphone/luadialplan] Fixed Bug#0000574: After adding a new user the state of this user is "unknown".
* [locaphone/core] added function to read version from packaging version file
* [locaphone/gui] added configurable product name
* [locaphone/gui] replaced product icon, favicon and default CSS
* [locaphone/provisioning] removed obsolete snom firmware download scripts
* [locaphone/core] removed outdated extstated
* [locaphone/luadialplan] removed superfluous dialplan scripts
* [locaphone/luadialplan] Fix: If you forward a call to parallel call target then the the channel will be busy if one of the parallel call extensions is busy.
* [locaphone/core] Fixed paths in crontabs
* added product logo in configuration
* added penlight MIT-License file
* added git ignore files
* added replace function in gsdb
* fixed problem with empty filename on vmqueue
* added comment block for complex timer work
* fixed problem with calltimers and ivr
* added call completion files, re-enabled clearForwarders (addition to previous commit)
* added missing callcompletion core file
* changed error log line to debug in callcompletion
* changed exttype app.hangup to Channel:hangup
* fixed problem with returning calls on busy without checking for postdial
* [locaphone/luadialplan] Fixed typo and null pointer error in clip service.
* Changed SIP User-Agent and SDP Session Owner
* [locaphone/gui] removed information from help, option is not implemented anymore]
* [locaphone/core] removed comment from permissions.lua
* [locaphone/luadialplan] fixed typo in callcompletion which causes bug
* [locaphone/luadialplan] fix for queue->queue->ivr->ext bug
* [locaphone/provisioning] provisioning helper for KIRK-DECT
* [locaphone/database] database structure changes for KIRK-DECT provisioning
* [locaphone/core] configuration file changes for KIRK-DECT provisioning

gemeinschaft4.0 (3.90.1-precise1) unstable; urgency=low

* New upstream version

* LocaNet branch git/svn changes/additions since 3.1.3:

- [locaphone/voicemail] Fixed printing the wrong voicemail box extension in e-mail notifications
- [locaphone/sipgateways] Added the possibility to select the source of the target extension (invite or to-header) for each sip gateway.
- Refactored complete AEL dialplan and associated (AGI) scripts to LUA, refactored affected GUI parts, refactored call forward system
- [asterisk] Do not load Radius CDR module to get rid of "rc_avpair_new: unknown attribute" syslog message. (cherry picked from commit 1bbd77f56f4041105445657de5300b6621535808)
- [locaphone/provisioning] Added new feature for snom phones: Show date/time stamp for all dial-log entries instead of only time stamp for present-day calls and date stamp for earlier calls.
- [locaphone/fax-integration] Updated documentation for hylafax server > version 6 and t38modem.
- [locaphone/admin] Changed default value for directmedia to no.
- [locaphone/agentmonitor] Fix: More javascript fixes for Mozilla Firefox >= 18.0 (cherry picked from commit ef8bed2b0f0903640b11e824a5997b455960712c)
- [autoprovisioning/snom] Provision telephone-event for FW >= 8.7, fixes non-working DTMF input
- Make preferred language selectable for the user
- Setter function for user language

LocaPhone VoIP TK-System Release 3.2

gemeinschaft3.0 (3.2.0-precise1) unstable; urgency=low

* New upstream version

* LocaNet branch git/svn changes/additions from branch locaphone-3.1 to locaphone-3.2:

  - [locaphone/monitoring] Added peer monitor.
  - [locaphone/agentmonitor] Added showing the agents penalty
  - [locaphone/agentmonitor] Added info that the agent is in wrapup time.
  - [locaphone/phonebuttons] Added queue callforward button
  - as last year mentioned, --user parameter is incompatible with --defaults-extra-file, so removing it from file
  - [locaphone/provisioning] Added new feature for snom phones: Show date/time stamp for all dial-log entries instead of only time stamp for present-day calls and date stamp for earlier calls.
  - [locaphone/fax-integration] Updated documentation for hylafax server > version 6 and t38modem.
  - [locaphone/admin] Changed default value for directmedia to no.
  - [locaphone/agentmonitor] Fix: More javascript fixes for Firefox >= 18.0
  - [autoprovisioning/snom] Provision telephone-event for FW >= 8.7, fixes non-working DTMF input
  - Make preferred language selectable for the user
  - Setter function for user language
  - [gemeinschaft/dev] Improved .gitignore for easier development and testing.
  - [gemeinschaft/gui] fix: Removed extension modules from provisioning/softkey-profiles for snom 720/760 phones.
  - Remove obsoleted README
  - Remove obsolete SVN diff script
  - Added button `WaitingQueueCallers` for Aastra phones.
  - Added an exit key to leave the XML menus of snom phones instantly.
  - Quickdial prefix is now editable via the web gui (external phonebook). Little improvements.
  - Improved agent and queue monitor. The agent monitor is now available with smaller icons
  - Added script and crontab to create automatic backups
  - Improved queue-monitor2. Calls can be grouped by the incomming did.
  - The function key `dial log` can be controled by the astbuttond for snom phones.
  - New phone button type `Queue Caller Waiting` for snom phones.
  - Added notification if we are not allwed to observe any queue. If you are not allowed to move agents, the agents sidebar is not visible.
  - Fixed alarm function: The username and the extension might not be the same.
  - Added alarm notification system.

LocaPhone VoIP TK-System Release 3.1

gemeinschaft3.0 (3.1.3-precise1) unstable; urgency=low

* New upstream version
* Source repo update
* Close Bug#539

* LocaNet branch git/svn changes/additions for branch locaphone-3.1:

  - [dialplan] Fix: Don't do extensive BUSY handling (e.g. forwards) when user is called from a huntgroup
  - [locaphone/agentmonitor] Fix: Mode buttons disappeared when changing the mode (bugfix for Firefox >= 18.0)
  - [dialplan-scripts] Fix: Compare huntgroup-ext as string to fix wrong extension matching
  - [dbupgrade] Add missing foreign key update for 'routes' table
  - [asterisk/ami] fix: Originate permission for PhoneSuite dialing via AMI
  - Added missing queue_log events to snom and Aastra XML applications
  - Improved privileges for ami user `astbuttond`.
  - Custom extensions in context from-internal-custom may now be called across cluster nodes
  - BUG#0000538: Replaced wrong variable $t in opt/gemeinschaft/htdocs/prov/siemens/dial-log/dlog.php (by patch)
  - BUG#0000547: Fixed enter/leave conditions for queues.
  - BUG#0000536: Added missing $ in pb_on_phone.php
  - BUG#0000543: Fixed searching for agents by number.
  - BUG#0000545,0000546: Notify the astbuttond that a queue has been modified or deleted.
  - Disabled choosing the native blf function key for snom phones.
  - Improved configuration for snom firmware 8.7
  - Fixed clip-function. One digit numbers are now selectable via the feature code.
  - Group-Pickup: Do groupcount (fixes issues with call waiting)
  - When calling a queue, do not send connected line updates that would expose the name of the agent to the caller
  - Removed reference to Gemeinschaft from lost password email
  - Renamed database upgrade SQL files to follow a consistent naming scheme
  - Resize SIP gateway username to 50 characters in asterisk.sql
  - Bugfix: removed AMI action from agent logon script because it results in duplicate logons (1 x realtime agent + 1 x dynamic agent)

gemeinschaft3.0 (3.1.2-precise1) unstable; urgency=low

* New upstream version

* LocaNet branch git/svn changes/additions for branch locaphone-3:

  - The user should stay in the queues even if he logs off from his phone.
    Fixed function of commit 9f04db3681353772252a190e4c4d640442f38313:
    New userevent to notify that a channels has been forwarded (directly)
    to another extension.
  - Added additional queue monitor.
  - Added now tab for the agent monitor.
  - Added missing image for agent monitor
  - prov-aastra: Let CT variants of phones get recognized correctly
  - prov-aastra: Make provisioning and phone menus i18n-aware
  - ldap-integration/install-openldap.sh: Set BDB flag to auto-cleanup
    transaction logs
  - ldap-integration: Script to re-initialise/recover the directory after
    e.g. unrecoverable LDAP DB failure
  - Added missing values to example queue in asterisk.sql
  - Disallow pressing agent pause/unpause function keys more than once
    every 5 seconds to avoid excessive Asterisk realtime database updates.
  - Aastra: allow agent to unpause even if the agent is not paused in all
    queues.
  - Generate random AMI Action IDs when sending AMI Actions
  - Pause and unpause agents in database and not only via AMI Action just
    to be sure it really happens in the database
  - Extended script gs-agent-logoff so that only agents whose users are
    not logged on to any phone may be logged off
  - Fixed typo that prevented quickdial help from being shown in Web UI
  - Fixed Javascript locale preloading in live monitor UI
  - New AMI wrappers for QueueAdd and QueueRemove actions
  - snom/Aastra: send QueueAdd AMI action when logging on agent to ensure
    that Asterisk definitely knows what queues the agent is logged on to.
    This fixes a race condition which resulted in agents not being paused
    in all queues when using autopause and/or autopausehangup

gemeinschaft3.0 (3.1.1-precise1) unstable; urgency=low

* Import to precise repository

gemeinschaft3.0 (3.1.1-lucid1) unstable; urgency=low

* New upstream version
* Remove annoying robot-dity moh
* Update conffiles to correspond with changed filelist in /o/g/e/asterisk/
* Fix accidential creation of /usr/share/doc/gemeinschaft/*
* Remove dh_compress so contents in doc/gemeinschaft3.0-core/ won't get
  gzipped anymore

* LocaNet branch git/svn changes/additions for branch locaphone-3:

  - Call completion can be switched on/off in gemeinschaft.php
  - Changed order in which snom phones are being shown in web GUI
  - Added name of Asterisk configuration parameters to user administration page
  - Added weight queue parameter to web GUI
  - Turned off CANONIZE_OUTBOUND and SAMSUNG_PROV_ENABLED if not
    configured in gemeinschaft.php
  - More reasonable defaults in gemeinschaft.php
  - Provisioning overview: More Polycom SoundPoint phones, add
    SoundStation conference phones
  - Polycom: use persistent audio volume across calls
  - New userevent to notify that a channels has been forwarded (directly)
    to another extension.
  - Fixed dialplan issue that prevented users from checking foreign
    mailboxes if the foreign mailbox is the only mailbox
  - Moved settings menu key to arrow right key in idle screen of snom
    phones.
  - Fixed a major bug in the privat phonebook search of smasung and
    polycom
  - Improved key settings for firmware 8.7 and snom-7xx phones
  - Improved gui for snom-760 softkey settings.
  - snom phones: removed ringer animation to gain more space for name and
    number display
  - Aastra phones: make XML application post list user-writeable to
    improve PhoneSuite compatibility
  - Increased maximum queue timeout from 250 to 65535 seconds
  - Change active value of autopause from "yes" to "all" to avoid problems
    with agents in multiple queues
  - Improvements and enhancements for billing module (calculation based on
    netto prices; options for shortened number output and display of
    failed calls, option to specify billing interval; configurables moved
    to global config file)
  - New feature: optional annoucement to the caller if number of callers
    waiting exceeds a given value
  - Allow configuration of autopause queue parameter via Web UI
  - Removed call forward dialplan hints
  - Removed deprecated call file generation to toggle call forward LED on
    some phones

- Also lookup local usernames in out-get-callername.agi
- Macro to update CONNECTEDLINE vars to send displayname information to calling phones
- Changed default value for sendrpid to 'pai' for sip peers.
- Fixed softkey hold for firmware 8.7.
- Added snom-710 support.
- Fixed problem with transfering calls that passed a huntgroup to normal extensions (call forwarding on no answer did not work).
- Added support for snom 8.7 firmwares.
- Fixed issue with intercom calls and group count.
- Fixed synchronisation of external phonebook to LDAP server
- Fixed callback in cluster mode
- Updated provisioning for snom firmwares newer than 7.4.33.
- Added comments for the ape-service to the gemeinschaft.php.
- Make gs-ldap-phonebook-sync more robust when importing duplicate external and private phonebook entries
- New feature: optionally keep the ringtone of the original callee when forwarding a call to another user.
- Added rpid-settings to user and sip-gateway configuration.
- Fix: Agent login/logoff script did not stop on errors.
- Fixed configuration for ape service in gemeinschaft.php

gemeinschaft3.0 (3.1.0-lucid1) unstable; urgency=low

* New upstream version
* Update LocaPhone branding patch to match APE server additions

* LocaNet branch git/svn changes/additions for branch locaphone-3:

- Anonymous calls can be forwarded correctly now
- Added new permission to `allow modifying` the agents queues.
- Added mananger events vor changing group permission states and agents queue memberships.
- Added new manager events for the permission system and the agents queues
- Agents will not need to log out and in to make changes active on their queues.
- Two manager events had a wrong name.
- Added manager events for adding/removing agents.
- Added live monitor for agents.
- Added multi language support, fixed bugs and improved the behaviour of the client script for the agents live monitor.
- If we remove a logged in agent, we will also remove him from the queues
- Fixed typo, compiled mootools
- Prepared the livemonitor to use SSL
- Improved communication between astbuttond and ape-server. Improved gui.
- Autopause on hangup should pause an angent in all queues.
- Improved ape-script. Fixed read errors.
- Added new function `clip only for the next call`
- Snom phones are using an actionurls for setting clip function now.
- Added new permission group of type agent.
- New Queue-Monitor. Improved Agent-Monitor.
- Improved styles. Added images. Added container for new icons in the queue monitor.
- Improved the gui of the queue monitor.
- Added missing images.
- Added support fo another expansion module for snom phones
- Added support for snom 720, snom760, snomMP
- The system will recognize that the secretary is secretary (by permission 'override_callforward_call') and she does not need to dial any prefix to call the boss anymore.
- We do not want to set the voicemail system language to german statically.
- Removed not working code.
- Voicemail notifications have not been translated.
- Added missing new columns in table route_in.
- Added missing id for menu entry billing
- Add Samsung Support
- initial implementation of extesion modules in keyprof file
- fixing display issue for samsung extension modules
- initial implementation of extesion modules in keyprof file
- fixed the ordering of the back and exit buttons
- fixed the ordering of samsung back and exit buttons
- fixed the keycount for samsung AOM modules
- ignore samsung firmware folder
- fixed ringtome provisioning for samsung phones, added localization to error message of gs_ringtone_set.php
- fixed the localization issue caused by merging
- fixed the umlaut issue in diallog localization with the hlp of utf8_decode (really dirty solution) needs to be fixed whe samsung supports UTF8
- added translation to function keys menu for samsung
- fixed the menuentry name for samsung function keys in webgui
- Add Link to key-layout pdf in samsung Softkeys
- Add Samsung to key-layout pdf-generation file
- Remove link to smt-i5230 key-layout-pdf
- First attempt to configure Aastra OMM DECT
- Implemented isConnected() in MySQL YADB driver
- Aastra OMM DECT provisioning incl. hotdesking (via external application)
- Include context to-internal-custom in default context to allow
- Bugfix: keep sort order in user administration Web GUI
- Do not sync hidden users with LDAP phonebook server
- Changed background color of snom key labels
- Fixed AASTRA_PROV_OMM_ENABLED switch in auto-provisioning script
- Added support for private phonebook to local LDAP server
- Added inbound routing by date

LocaPhone VoIP TK-System Release 3.0

gemeinschaft3.0 (3.0.5-lucid1) unstable; urgency=low

* New upstream version

* LocaNet branch git/svn changes/additions for branch locaphone-3:
  - Anonymous calls can be forwarded correctly now
  - snom: changed call forward function keys to action URL
  - Added inbound routing by date
  - Removed debugging output

gemeinschaft3.0 (3.0.4-lucid1) unstable; urgency=low

* New upstream version

* LocaNet branch git/svn changes/additions for branch locaphone-3:

  - Updated/complete english string translation
  - Fixed logging format in fake QueueLog() calls
  - Log disabled Aastra provisioning as WARNING instead of DEBUG

gemeinschaft3.0 (3.0.3-lucid1) unstable; urgency=low

* New upstream version
* Updated snom manual

* LocaNet branch git/svn changes/additions for branch locaphone-3:

  - Cache phonetype also on *0*-logout
  - Aastra: fixed phone type cache handling when logging off a user from the phone.
  - Aastra: improved message text if agent is already logged on
  - snom: moved generic XML functions and error message display into inc/snom-fns.php
  - snom: Allow phone to be reconfigured without reboot if no reboot is necessary
  - snom: hotdesking of users via minibrowser
  - snom: queue agent logon/logoff and agent pause via minibrowser
  - Do not send duplicate pause/unpause AMI events

gemeinschaft3.0 (3.0.2-lucid1) unstable; urgency=low

* New upstream version
* Updated phone manuals with fixed authors

* LocaNet branch git/svn changes/additions for branch locaphone-3:
  - The userevents for (un)pausing an agent are not needed anymore. PauseQueueMember and UnpauseQueueMember are doing their work now.
  - Fixed playback of alternative announcement file before voicemail on external calls
  - fixed misplaced closing bracket in internal HylaFAX authfile function
  - Enabled phonebook reverse search for queue calls
  - Added database upgrade script for post v3 changes
  - snom: fixed hiding of hidden users in internal phonebook
  - Do not remove CALLERID(num) when hiding the caller-id
  - Do not include context to-internal-custom in from-internal-users

gemeinschaft3.0 (3.0.1-lucid2) unstable; urgency=low

* Remove quilt include from rules and control file, add source/format

gemeinschaft3.0 (3.0.1-lucid1) unstable; urgency=low

* New upstream version
* Updated Snom user manual
* New user manual for Aastra phones

* LocaNet branch git/svn changes/additions for branch locaphone-3:
  - removed deprecated script version
  - moved agent-logoff-script into scripts directory
  - Prevent duplicate P-Asserted-Identity and P-Preferred-Identity headers
  - Aastra: workaround for broken voicemail hardkey on 6739i
  - Aastra: "sip vmail" URI is required for other models, too
  - Changed ringtone labeling in Web GUI
  - Fixed typo in database upgrade query of table gate_params
  - snom: fixed typo in language version array
  - Removed duplicate functionality which creates hint helper callfile (also in inc/gs-fns/gs_callforward_activate.php)
  - Fixed some undefined variable PHP warnings
  - removed superfluous script variables
  - Aastra: improved DND debugging
  - New Aastra language files
  - more useful defaults for snom provisioning
  - Aastra: placeholders in key profiles are now being correctly replaced
  - Enable group pickup on Aastra phones

gemeinschaft3.0 (3.0.0-lucid1) unstable; urgency=low

* New upstream version tag, no code changes
* Updated music on hold, updated moh-install file

gemeinschaft3.0 (2.90.3-lucid1) unstable; urgency=low

* control: asterisk-config-gemeinschaft3.0 now conflicts with
  asterisk-config, seemingly fixes automatic dependency resolving
  on "apt-get install gemeinschaft3.0".
* Remove e-emergency.ael.php from -core.conffiles (doesn't exist anymore)

* LocaNet branch git/svn changes/additions for branch locaphone-3:
  - For asterisk 1.8 the astbuttond also needs the privilege `command`
  - The `o`-Option for the dial applikation seems not to work anymore in
    asterisk 1.8, so we can remove this
  - The aastra 6739is do not have topsoftkeys, so we will not configure
    them. Moreover we can configure 10 more softkeys
  - If no number has been send the callerid will be set to `anonym`
  - Added missing script for pausing aastra phones
  - Fixed dialplan for asterisk 1.8. The extension matching behavior is
    different from previous versions.
  - We will not playback the announcement if we will not enter the queue.
  - Column `ipaddr` in table `ast_sipfriends` needs to be larger (45 chars).
  - Renamed column `_uniqueid` to `uniqueid` in table `ast_voicemail` (for
    asterisk 1.8)
  - Added link to snom manual for snom 820 and 821
  - Fixed table `ast_voicemail`
  - Resized column in table (for asterisk 1.8)
  - Renamed `username` to `defaultuser`. We will not print empty options to
    the sip gateway config options anymore
  - Allow column being null in table
  - Commit (basic) billing module
  - Add "support" for Asterisk 1.8 CDR entries
  - Log extension instead of username into CDR accountcode field when doing
    private calls
  - MySQL: View for all internal LocaPhone users
  - Small number presentation fix for Asterisk 1.8
  - MWI subscription changes for Asterisk 1.8
  - Make sure SIP registration is in [general] section of sip.conf
  - Removed emergency call configuration via gemeinschaft.php. Use routes
    and LCR instead.
  - snom: increase subscription expiry to 600 seconds to reduce system load
  - e-emergency.ael.php is no longer needed.
  - Fixed typo in Set(CALLERID(num-pres)...
  - Changed queue-answer Macro into GoSub context to work around Asterisk
    1.8
  - Do not apply GROUP_COUNT() twice on forwarded internal calls
  - Improved help text for CSV phonebook import
  - Added more language versions to snom provisioning
  - removed duplicate call box
  - Disabled explicit MWI subscription to prevent Asterisk warning messages
    from
  - snom: do not show advertisement in Web UI
  - Added support for exporting phonebooks from MySQL database to local LDAP
  - Added script author to boilerplate

gemeinschaft3.0 (2.90.2-lucid1) unstable; urgency=low

* LocaNet branch git/svn changes/additions for branch locaphone-3:
  - Fixed hack for inbound calls from sipgate.de and extended function to
    work with Toplink and QSC, too.
  - Fixed group permissions check for LDAP user lookup service.
  - Use ldap_search() instead of ldap_list to allow LDAP subtree searches.
  - changed refresh interval for iaxmodem registration because old value of
    0 breaks iaxmodem 1.1.1
  - Fix polycom minibrowser default page/application
  - Recommitted quickdial function
  - Fixed pamal LDAP module for LDAP Web GUI authentication.
  - Escape HTML special characters in Polycom phonebook.
  - Fixed some bugs in the callforwarding funktion for aastra phones and
    softkeys.
  - Fixed logging in an agent on an aastra phone while this agent is already
    lofed in on another phone.
  - Fixed deleting items from the dial log via the the xml menu of the
    aastra phones.
  - If we use the AstButtond the Aastra phone does not need to count missed
    calls.
  - The permission 'login_queues' does not exist anymore.
  - SIP configuration changes for Asterisk 1.8
  - Aastra provisioning: remove 'options password enabled' to enable access
    of headset settings via options menu.
  - Set CDR accountcode for outgoing calls.
  - Fixed and improved sip gateway configuration. We can now configure
    custom parameters again.
  - Added example config files for starting/stopping iaxmodem and faxgetty
    with initctl.
  - In the dial log for missed and answered queue calls on the snom phones,
    we do not need to add the information that the call belonged to a queue.
  - The dial log of the aastra-phone grouped missed and answerded queue
    calls only by number. Now it also groups the calls by queues.
  - Fixed deleting calls from the dial log via the snom xml menu.
  - The lower buttons on the aastra 57i ar not configurateable for users.
  - Set CDR userfield on private calls

- Do not compare integers as strings when using GROUP_COUNT()
- Fixed call groups and GROUP_COUNT() counting
- aelparse is no longer available, remove opt/gemeinschaft/sbin/gs-ast-ael-check
- Fixed function gs_callforward_activate.
- Fixed cdr gui so we can now also see outgoing calls again.
- Added pause button for queue agents on aastra phones
- Fixed call counter on parallel call and boss/secretary function
- Fixed setting DIALSTATUS=busy when calling local users

gemeinschaft3.0 (2.90.1-lucid1) unstable; urgency=low

* Gemeinschaft/LocaPhone 3.0.0beta1 - THIS IS CONSIDERED BETA-SOFTWARE, YOU WILL EXPERIENCE PROBLEMS! USE AT YOUR OWN RISK!
* Package based on gemeinschaft2.0-2.3.2.4-lucid1

* LocaNet branch git/svn changes/additions for branch locaphone-3:
  - Gefixt: Tabelle queue_callforwards, queue_cf_timerules, cf_timerules fuer die neuen Rufumleitungssachen, es haben div Spalten gefehlt, weswegen die Rufumleitung nicht funktionierte. uebernommen
  - Datenbank-Upgrade fuer commit 7bb235801be4c4169b9f2383e14f909358d86a33 hinzugefuegt.
  - DB-Aenderungen aus commit 415eb2bc76d9b3dd6e4167d6bfdb0ef21b992dd0 uebernommen
  - Tabelle ast_sipfriends und den View ast_sipfriends_gs fuer asterisk 1.6 angepasst.
  - Die Funktion conv_ringtone hat nun zwei Parameter.
  - Added version info to the gui
  - Received and missed queue calls will be displayed in separate menues
  - The dial logs for queues are visible by default.
  - Improved the look and feel of many lists in the gui
  - Fixed typo
  - We can now select the internal ringers in snom phones.
  - permission name is `intercom_call` and not `intercom`
  - All calls via the gateways are external calls. It does not make sense to restrict the trunk prefix to `0`.
  - Der DND-Status wurde in eine eigene Tabelle ausgelagert.
  - Die Datei euvg.conf sollte eigentlich uevg.conf heissen.
  - Der Gruppen-Pickup funktionierte nicht.
  - Wurde ein Anruf in eine Warteschlange umgeleitet, so funktionierte das Erfassen im Diallog nicht oder nicht korrekt.
  - Beim Aendern der Sprache wird ein UserEvent generiert.
  - Fehler beim Mergen der Sprachunterstuetzung fuer Polycom beseitigt
  - Fixed dnd-Function for Polycom phones
  - Der Username konnte nicht anhand der Extension bestimmt werden.
  - Script aktualisiert
  - Bug#247: Man kann den automatischen Rueckruf bei besetzt auch ausloesen, wenn die Ansage noch abgespielt wird.
  - Der DND-Status wurde in eine eigene Tabelle ausgelagert.
  - Beim aendern des DND-Status und nach dem Aufrufen der Verpassten Anrufe wird auch fuer Aastra Telefone ein UserEvent im Asterisk erzeugt.
  - Ueberfluessige Zeile `dnd` aus der Tabelle `users` entfernt.
  - An den Polycom Endgeraeten konnte nicht DND aktiviert werden.
  - Beim Ein-/Ausloggen eines Users ueber das Aastra-XML-Menue werden UserEvents erzeugt.
  - Vergroessern der Spalte `mac_addr` fuer die Tabelle `phones`, sodass die Provisionierung fuer das Snom M3 funktioniert.
  - gs_user_missedcalls_ui benoetigt die Extension als Parameter und nicht den Usernamen
  - The system will check the group permissions when pressing the dnd-key on Aastra phones
  - Fixed searching imported in snom phonebook
  - Forwarded calls will be assigned to the last user who forwarded the call.
  - Some strings needed to be HTML-Encoded
  - Added agent logoff script
  - Improved support for aastra phones. Added softkeys.
  - Fixed wrong diversion target in the dialplan.
  - The login screen did not support long usernames.
  - Call completion will also work with parallel call diversion
  - Improve usability on SPIP32x/33x phones by providing a mainmenu-softkey on the idle display
  - README for firmware deployment
  - Internal phonebook entries get displayed based on phonebook_user permission(s)
  - Initial i18n work - user selectable phone language preference (missing dial plan additions)
  - AGI language helper, store default language for peer on creation
  - get-user-language modified to always return language in both formats in two separate vars
  - get-gs-language-by-ast-code dialplan helper - convert 2 letter asterisk language to gs/iso language
  - Changes to dialplan for i18n - may need further work
  - I18N enhancements for Polycom provisioning
  - Make user_add and user_change functions in inc/gs-fns/ language-aware
  - Missing bits for gs-users-get and gs-users-regen regarding i18n support
  - GIT merge comments fixed ; Updated XML applications for Polycom, e.g. call forwards and group permission handling ; Fixed display of parallel and timeconditioned call forwards in XML applications for Polycom and Snom
  - Make rest of Snom XML applications I18N-aware
  - Missing Gettext tags

- Add missing getuserid for permission check; minor cleanups
- Bug 478 - Playback cc-offerings in user's language
- Bug#480: Check IP/MAC permission on dial-log invocation for polycom phones
- Bug#480 additionally fixed for snom aps
- s/&/& to play nice with polycom
- When doing phone_checkcfg, try to determine target phone type and call that checkcfg-mechanism only.
- Improve checkcfg phonetype detection mechanism
- Add prov_checkcfg support to Aastra XML login application, typo
- prov_checkcfg: revert database hack, implement and use phone type caching mechanism for proper phonetype-dependent reboot/recheck when using the roaming feature
- Weitere Verbesserungen Aastra Autoprovisionierung
- aastra.cfg brauchen wir nicht mehr
- Erweiterungs-Module an Aastra-Endgeraeten
- snom m3 Provisionierung angepasst
- Menuepunkt "Weckruf" in Hauptmenue "Administration" verschoben.
- Benutzerliste in der GUI kann nach Benutzername, Name, Nebenstelle, Email und Host sortiert werden
- snom m3 Provisionierung: Benutzername/Kennwort fuer Zugriff auf Basisstation
- snom m3: Benutzername am Handgeraet richtig anzeigen, Basisstation mit HTTP-Passwort rebooten
- Zum Ermitteln der IP-Adresse des Telefons muessen wir nur die Slave-Datenbank abfragen
- Loggen, wohin das Callfile wirklich verschoben wird
- Bug 475 behoben: Spalte id in Tabelle queue_cf_timerules hinzugefuegt, damit Zeitsteuerungseintraege bearbeitet werden koennen
- Bugfix fuer snom Tastenbeschriftung
- Bei den Beistellungen der Aastra-Telefone konnte man eine Tasten mehr Konfigurieren als eigentlich vorhanden war.
- snom m3 Provisionierung: Benutzername/Kennwort fuer Zugriff auf Basisstation
- In Firmware 2.6.0.1008 fehlender Standard-Dialplan wird jetzt provisioniert.
- Fehler bei der Provisionierung leerer Tasten behoben.
- Bugfix fuer Anzeige von Aastra Beistellungsmodulen in Web-GUI
- Gruppen koennen jetzt Parameter haben
- New features in imported phonebook:
  - items can be edited if user group has permission 'phonebook_imported_edit'
  - items can be added if user group has permission 'phonebook_imported_edit'
  - visibility of items can be restriced to user groups
  - dialplan-scripts/in-get-callername.agi honors group restrictions
- Items in imported phonebook should be visible to all users by default
- Apply group permission restrictions on imported phonebook items
- Increased length of column type in table group_permissions to 25
- Fixed bug on Aastra phones that caused inherited expansion module keys
- Added missing column directmedia to table ast_sipfriends and to view
- Added missing column directmedia on table ast_sipfriends
- Fixed supervisor in ast_sipfriends table
- Fixed expansion module key numbering on Aastra 53i.
- New AMI events QueueLoginUI and AgentLoginUI
- Added AMI user event
- New agent key and call forward key on Aastra phones
- Added database upgrade statements for tables group_parameters and pb_ldap
- Cleaned up and improved sample HylaFAX configuration files
- Reduced minexpiry to 60 seconds to match SUBSCIBE timer
- Fixed searching in snom phonebook
- Added missing call waiting AGI script.
- res_timing_pthread und res_timing_timerfd deaktiviert.
- Do not show called account name on incoming calls on Aastra phones
- Use column 'pause' of database table 'ast_queue_members' to determine agent
- Added missing queue configuration parameters.
- Set periodic_announce_frequency to 0 if announce_frequency is 0
- Use Asterisk dialplan application PauseQueueMember()/ UnpauseQueueMember() to pause and unpause agents
- Minor enhancements to Siemens provisioning installation instructions.
- Added external number prefix "0" to numbers inserted from drop-down list on
- Removed superfluous Aastra dialplan parameter
- Added support for OpenStage 15 incl. expansions module (sidecar)
- Siemens: Preload dial log and phonebook applications
- Siemens: Cleanup and minor changes for firmware 2.1
- New internal function to send push request to Siemens phone.
- Added configuration option FAX_NOAUTH_LOCALHOST to allow local connections
- Enabled missed calls counter on Aastra phones.
- Erweiterungs-Module an Aastra-Endgeraeten
- Always turn on call waiting on snom phone because call waiting is now being handled on the asterisk server
- Use Debian default SSL file locations in example.
- Allow user to subscribe to more than one mailbox. If subscribed to more than
- Added Queue() callback macro to set GROUP() count properly if an agent answers the call. Otherwise, incoming calls to the agent's extension will not honor the call-waiting settings.
- Changes to web-based voicemail playback:
  - Added OGG/Vorbis to the list of supported Voicemail output formats.
  - Replaced plugin-based audio playback for browsers that support HTML

<audio> tags (Firefox >= 3.5 required, Safari).
- Removed outdated code from vm-play.php.
- Try to guess output format from HTTP User-Agent
- New download URLs for Aastra firmware
- Initial support for Aastra 6730i, 6731i, 6739i and Aastra firmware 3.2.0
- Record calls using the record button on snom phones.
- Fixed display of caller-id from P-Asserted-Identity headers on Aastra phones.
- Fixed recording of outbound calls and record filename when using the snom
- Aastra: Fixed internal phonebook, added support for external (imported)
- Aastra: added missed calls counter to missed calls list, added ability to delete calls from call lists.
- Prefix description modified
- Dialstring for "DAHDI" added.
- use prefix on DAHDI channels
- Routing test displays group
- Permission type "record_call" added
- Database changes preparing realtime monitor integration
- Realtime Monitor client side
- get callforwarding by user ID
- QueueMon updated
- PeerMon added
- Realtime Monitor Graphic Lib updated
- display comment field on home page
- user call-init.php as default
- database changes for realtime monitor
- display more than 7 rows in group/queue monitor
- added sequence number to messages
- Vodeo codecs added.
- Division by zero fixed.
- Create empty members array on empty queue members table.
- Modifications to Silver Bullet client library and GUI modules
- HTML encoding improved
- Change extensions state to "unknown" on connection errors
- Initial Silver-Bullet commit to Gemeinschaft Repo
- Silver-Bullet config changed: do not daemonize
- SilverBullet logger fixed
- PeerMon Module added
- Fixed ignoring GS_GUI_NUM_RESULTS in fax GUI
- Added LSB tags to Silver-Bullet init script
- Silver-Bullet init script updated to launch SB after Asterisk starts
- Silver-Bullet: Delays in queue count fixed
- GUI_SUDO_EXTENDED set to "true" as defalt value
- https://bugs.launchpad.net/gemeinschaft-amooma/+bug/446990
- https://bugs.launchpad.net/gemeinschaft-amooma/+bug/553544

("Nachrichten auf dem AB per ISDN werden als von anonym angezeigt")
- Alle in RFC 3261 erlaubten Zeichen in SIP-Benutzernamen auf Gateways erlaubt (https://bugs.launchpad.net/gemeinschaft-amooma/+bug/485413)
- `/etc/init.d/asterisk start` statt `safe_asterisk`
- set_magic_quotes_runtime() is deprecated
- VPB-Treiber auswhlbar gemacht
- "get_class() expects parameter 1 to be object, array given in lib/getopt.php on line 242 (suppressed)"
- noload => res_config_sqlite.so
- noload => res_config_pgsql.so
- noload => res_config_ldap.so
- jingle.conf
- noload => cdr_sqlite.so
- users.conf auskommentiert
- Beispiele in cdr_mysql.conf
- res_mysql.conf, cdr_mysql.conf: weniger Warnungen
- "Lauter, härter, schneller" aus ChangeLog entfernt
- locale: .pot aktualisiert
- locale: update-po
- locale: make gen-mo
- locale: Do not use "/../inc/conf.php" (which would load "/etc/gemeinschaft/gemeinschaft.php") on the build system.
- locale: make gen-php
- locale: make gen-php
- s/Authorisierung/Autorisierung/
- falsche Verwendung von __() gefixt
- "0" muss hier nicht übersetzt werden
- Queue-Extension muss mindestens 2 Ziffern lang sein
- Leere index.html-Dateien mit irgendeinem Inhalt gefuellt.
- voicemail.conf: pollmailboxes=yes
- listen-to-ip rausgenommen
- Anpassung f. XML Pickup Info Asterisk 1.6
- Asterisk 1.6 Anpassungen
- Mysql Fehlermeldungen beseitigt (teilweise)
- Diallog Queues braucht MEMBERINTERFACE
- Provisioning f. twinkle
- BUGFIX: sox parameter changes
- Enable silver bullet monitors by default
- BUGFIX: error message in log (diallog queues)
- Callwaiting provisioning. Rufumleitung am Telefon deaktiviert.
- Gruppen-Funktionen: Array-Inhalte untereinander (bessere Lesbarkeit)
- Intercom mit Ueberpruefung der Berechtigung hinzugefuegt
- IAX-Gateways hinzugefuegt (https://bugs.launchpad.net/gemeinschaft/+bug/420105)
- Grandstream 'autoanswer by call-info' fuer Intercom aktiviert
- Tiptel Provisionierung hinzugefuegt (https://bugs.launchpad.net/gemeinschaft/+bug/507480)
- Anpassung voicemail.conf an Asterisk 1.6.2.6

- Anpassung an Asterisk 1.6 SIPCHANINFO() ist deprecated, ersetzt durch CHANNEL()
- asterisk minivm.conf hinzugefuegt aus asterisk 1.6.2.6 config samples
- scripts/gs-group-connections-get hinzugefuegt
- dialoginfo fuer pickup abgeschaltet, damit Pickup mit den Tiptels funktioniert
- Polycom-Config nach Tiptel geaendert
- Warteschlangen koennen per GUI einer Gruppe hinzugefuegt oder geloescht werden. Wie bei den Users.
- Zeilenumbruch war an falscher Stelle
- Anzeige der Service Nummern Weckruf und Warteschlangen in der Hilfe nur bei Berechtigung
- Gruppensystem: Permission private_call hinzugefuegt
- GUI Weckruf in Gruppe admin_gui verschoben
- Neue Gruppe nobody_users hinzugefuegt
- Grandstream, Snom und Tiptel Telefoneintraege nur wenn Permission 'phonebook_user' auf eine Gruppe
- phonebook type 'gs' braucht $mac und $user
- Tiptel Prov.: Anzeige der Rufnummer hinter lastname, firstname (number)
- Ueberpruefung von $action verbessert
- Tiptel Prov: XML-Browser fuer PB und Dial-Log, LineKey2=XML-Telefonbuch und LineKey3=XML-Dial-Log
- Tiptel Prov: Suchen im XML-Browser PB eingebaut.
- Dial-Log -> Anruf Listen (Deutsche Anpassung)
- Beschreibung der Group-Permissions
- Tiptel Prov.: aktivate Intercom Config
- Voicemail mit Deutschen Ansagen, wenn weiterleitung auf Voicemail
- Tiptel IP28xs ins Provisioning hinzugefuegt
- add script gs-group-get (with --regen parameter)
- Anpassungen an andere gs-group* Files
- regen Parameter in gs-group-members-get und gs-group-permissions-get hinzugefuegt
- script gs-group-connection-add und gs-group-connection-del hinzugefuegt
- script gs-groups-regen hinzugefuegt
- GUI: Anzeige der Firmware-Dateien der verschiedenen Telefone
- DHCP Example Config for Tiptel Phones
- Tiptel XML Phonebook angepasst, kann max 15 Eintraege anzeigen. Mittels Buttons kann man Seiten ansehen.
- Tiptel XML kann nur max 15 Eintraege
- Tiptel XML Phonebook: Suchen Button hinzugefuegt
- Tiptel XML Phonebook: $url_snom_pb -> $url_tiptel_pb
- Tiptel Prov: Ausblenden der Menueeintraege 'Leistungsmerkmale' und 'Nachrichten' im Telefon
- Callwaiting fuer Caller; Vorher wurd nur EXTEN ausgewertet.
- callwaiting: &caller-count() fuer always forward und wakeup calls
- In sox version 14.0.1 ist der Paramter rate deprecated
- GUI-Einstellung Dienstmerkmal-Anklopfen (call waiting) fuer Grandstream Provisioning
- Grandstream Provisioning verbessert
- Bei eingestellten externen Klingelton wurde dem Grandstream bei einem Anruf von extern nicht signalisirt das der Externe Klingeton benutzt werden soll
- Imported Phonebook fuer das Grandstream Telefonbuch hinzugefuegt
- Grandstream GXP20x0 Tastenerweiterung 1 und 2 hinzugefuegt
- Anpassung an Firmware 1.2.2.19 (BT und GXP-Serie)
- Anpassung an FW 1.2.3.5 (BT und GXP Serie)
- GXP2010 kann auch XML Applikation ausfuehren
- definierte Ausgabe von $callwaiting
- New README.textile file.
- misc directory added
- Layout
- Links fixed
- Link to FAQ included
- Ueberschriften Sinnvoller fuer die Rufumleitungs-Konfiguration
- s/AB-Konfiguration/Konfiguration
- Umbenannt: gs_group* zu gs_prov_group* wegen Fehler in htdocs/gui/mod/prov_groups.php. Die funktionsnamen gs_group_add gs_group_change etc. wurden schon in der inc/group-fns.php benutzt.

LocaPhone VoIP TK-System Release 2.3

gemeinschaft2.0 (2.3.2.4-lucid1) unstable; urgency=low

* New upstream revision
* gemeinschaft2.0-core depends on vorbis-tools (closes bug#486)

* LocaNet branch git/svn changes/additions since 2.3.2.3-lucid1:
  - Cleaned up and improved sample HylaFAX configuration files
  - Removed superfluous Aastra dialplan parameter
  - Added external number prefix "0" to numbers inserted from drop-down list on VM configuration page.
  - Minor enhancements to Siemens provisioning installation instructions.
  - Some strings needed to be HTML-Encoded
  - New internal function to send push request to Siemens phone.
  - Siemens: Cleanup and minor changes for firmware 2.1
  - Siemens: Preload dial log and phonebook applications
  - Added support for OpenStage 15 incl. expansions module (sidecar)
  - Do not use local channel to forward calls if GS_INSTALLATION_TYPE == single
  - Always use local channel to forward calls from one user to another if parallel call is enabled. Using 'jump' will not work in this case.
  - Added agent logoff script
  - Added version info to the gui
  - We do not want the user to be logged out from queues if he logs out from a phone.
  - Do not use jump for callforwarding in cases it does not always work.
  - New download URLs for Aastra firmware
  - Initial support for Aastra 6730i, 6731i, 6739i and Aastra firmware 3.2.0
  - The login screen did not support long usernames.
  - Fixed wrong diversion target in the dialplan.
  - Improved support for aastra phones. Added softkeys.
  - Fixed recording of outbound calls and record filename when using the snom record key.
  - Added missing record_call group permission.
  - Added Queue() callback macro to set GROUP() count properly if an agent answers the call. Otherwise, incoming calls to the agent's extension will not honor the call-waiting settings.
  - Record calls using the record button on snom phones.
  - Fixed recording of outbound calls
  - Fixed recording of calls to other nodes on cluster
  - Changed permission checks of call recordings. Recording permissions can now be set based on host groups and user groups.
  - Aastra: Fixed internal phonebook, added support for external (imported) phonebook

gemeinschaft2.0 (2.3.2.3-lucid1) unstable; urgency=low

* New upstream revision
* Post-2.3.2.2 patches removed as they are contained upstream

* LocaNet branch git/svn changes/additions since 2.3.2.2-lucid2:
  - Bug#480: Check IP/MAC permission on dial-log invocation for polycom phones
  - Bug#480 additionally fixed for snom aps
  - Do not show called account name on incoming calls on Aastra phones
  - The system will check the group permissions when pressing the dnd-key on Aastra phones
  - Wrong log message. -> sox parameter changes
  - BUGFIX: empty dir for voicemail
  - Set periodic_announce_frequency to 0 if announce_frequency is 0
  - Added database table column and web GUI for queue autopausehangup
  - voicemail.conf: pollmailboxes=yes
  - Added missing queue configuration parameters.
  - Use column 'pause' of database table 'ast_queue_members' to determine agent pause instead of deprecated column of table 'agents'.
  - Use Asterisk dialplan application PauseQueueMember() and UnpauseQueueMember() to pause and unpause agents.
  - Forwarded calls will be assigned to the last user who forwarded the call.
  - Fixed searching imported in snom phonebook

gemeinschaft2.0 (2.3.2.2-lucid2) unstable; urgency=high

* Post-2.3.2.2 fixes for major bugs, already included in upstream GIT:
  - locaphone-2.3.2.2-callinit: Use call-init.php instead of call-init-2.php to dial from WebGUI
  - locaphone-2.3.2.2-fix-dbupdate: Fix/enhance post database upgrade
  - locaphone-2.3.2.2-fix-snompb: Fix broken Snom phonebook caused by implementation of group permissions
  - locaphone-2.3.2.2-meetme-opts: Remove silence/voice detection from Meetme() application calls
  - locaphone-2.3.2.2-missing-callwaiting-agi: Add missing AGI script
  - locaphone-2.3.2.2-timing-modules: Add noload for buggy timing modules to modules.conf which cause cutoff playback of system prompts (e.g. voicemail)

gemeinschaft2.0 (2.3.2.2-lucid1) unstable; urgency=low

* New upstream revision
* aastradefaults patch removed (file no longer used)

* LocaNet branch git/svn changes/additions since 2.3.2.1:
  - Bug 475 behoben: Spalte id in Tabelle queue_cf_timerules hinzugefuegt, damit Zeitsteuerungseintraege bearbeitet werden koennen
  - ALTER TABLE Statement repariert
  - Bugfix fuer snom Tastenbeschriftung
  - Beim aendern des DND-Status und nach dem Aufrufen der Verpassten Anrufe wird auch fuer Aastra Telefone ein UserEvent im Asterisk erzeugt.
  - Der DND-Status wurde in eine eigene Tabelle ausgelagert.
  - Verbesserung fuer Aastra Autoprovisionierung:
    + login.php benutzerfreundlicher gemacht
    + Login XML-Applikation wird auf 57i standardmaessig benutzt
    + Funktionstasten an 53i verfuegbar
  - Erweiterungs-Module an Aastra-Endgeraeten
  - aastra.cfg brauchen wir nicht mehr
  - Aastra Autoprovisionierung verbessert
    + Firmwareupdates
    + Aastra Sprachdateien hinzugefuegt
    + bessere Standardeinstellungen
    + SIP-Account und Tasten koennen ohne Reboot umkonfiguriert werden
  - Bug#247: Man kann den automatischen Rueckruf bei besetzt auch ausloesen wenn die Ansage noch abgespielt wird.
  - snom m3: Benutzername am Handgeraet richtig anzeigen, Basisstation mit HTTP-Passwort rebooten.
  - Fehlenden Primary Key hinzugefuegt.
  - snom m3 Provisionierung: Benutzername/Kennwort fuer Zugriff auf Basisstation und Anzeige der Nebenstelle am Handgeraet eingebaut.
  - An den Polycom Endgeraeten konnte nicht DND aktiviert werden.
  - Bei den Beistellungen der Aastra-Telefone konnte man eine Tasten mehr Konfigurieren als eigentlich vorhanden war.
  - Ueberfluessige Zeile `dnd` aus der Tabelle `users` entfernt.
  - Vergroessern der Spalte `mac_addr` fuer die Tabelle `phones`, sodass die Provisionierung fuer das Snom M3 funktioniert.
  - Beim Ein-/Ausloggen eines Users ueber das Aastra-XML-Menue werden UserEvents erzeugt.
  - In Firmware 2.6.0.1008 fehlender Standard-Dialplan wird jetzt provisioniert.
  - Bugfix fuer Anzeige von Aastra Beistellungsmodulen in Web-GUI
  - Fehler bei der Provisionierung leerer Tasten behoben.
  - gs_user_missedcalls_ui benoetigt die Extension als Parameter und nicht den Usernamen
  - Apply group permission restrictions on imported phonebook items
  - Items in imported phonebook should be visible to all users by default
  - New features in imported phonebook:
    + items can be added/edited if user group permission 'phonebook_imported_edit'
    + visibility of items can be restriced to user groups
    + dialplan-scripts/in-get-callername.agi honors group restrictions
  - Gruppen koennen jetzt Parameter haben, Gruppenbasierte Beschraenkung von maximal moeglichen Anrufen (Gruppen-Parameter call-limit)
  - callwaiting: &caller-count() fuer always forward und wakeup calls
  - Callwaiting fuer Caller; Vorher wurd nur EXTEN ausgewertet.
  - Increased length of column type in table group_permissions to 25
  - Changed listen-to-ip to 0.0.0.0
  - Fixed bug on Aastra phones that caused inherited expansion module keys to be empty.
  - Added missing column directmedia to table ast_sipfriends and to view ast_sipfriends_gs
  - Fixed supervisor in ast_sipfriends table
  - Fixed expansion module key numbering on Aastra 53i. Added call forward XML application for Aastra phones.
  - New AMI events QueueLoginUI and AgentLoginUI
  - New agent key and call forward key on Aastra phones
  - Do not provision non-existing softkeys on Aastra phones
  - Added missing 'break' in Aastra expansion module logic
  - Bug 478 - Playback cc-offerings in user's language

gemeinschaft2.0 (2.3.2.1-lucid1) unstable; urgency=low

* New upstream revision
* Patch fix-gsusernamebyext contained upstream and thus removed

* LocaNet branch git/svn changes/additions since 2.3.2.0:
  - Der Username konnte nicht anhand der Extension bestimmt werden.
  - Datenbank-Update fuer die Rufweiterleitung gefixt.
  - Datenbank geaendert, sodass nicht jeder jeden picken kann.
  - Script aktualisiert
  - GIT merge comments fixed
  - Updated XML applications for Polycom, e.g. call forwards and group permission handling
  - Fixed display of parallel and timeconditioned call forwards in XML applications for Polycom and Snom
  - Make rest of Snom XML applications I18N-aware
  - Updated en_US translation
  - Missing Gettext tags
  - Add missing getuserid for permission check; minor cleanups
  - new mode 100755
  - Boilerplate gefixt
  - gs_ami_events.php aufgeraeumt und gs_queue_logoff_ui() und
  - snom m3 Provisionierung angepasst

gemeinschaft2.0 (2.3.2.0-lucid2) unstable; urgency=high

* Patch fix-gsusernamebyext fixes integer-casting of usernames in inc/gs-fns/gs_user_name_by_ext.php to make astbuttond updated for missed calls (and maybe others) work again - already applied upstream, patch no longer needed in next version
* control: Depend on english prompts

gemeinschaft2.0 (2.3.2.0-lucid1) unstable; urgency=low

* New upstream revision

* LocaNet branch git/svn changes/additions since 2.3.1.3:
  - Die Datei euvg.conf sollte eigentlich uevg.conf heissen.
  - Der Gruppen-Pickup funktionierte nicht.
  - Wurde ein Anruf in eine Warteschlange umgeleitet, so funktionierte das Erfassen im Diallog nicht oder nicht korrekt.
  - Beim aendern der Sprache wird ein UserEvent generiert.
  - Uebersetuzng aus commit 5ed2190cad9e61c82a33170aa72262bcad4b10c3 uebernommen.
  - Fehler beim Mergen der Sprachunterstuetzung fuer Polycom beseitigt
  - Fixed dnd-Function for Polycom phones
  - Initial i18n work - user selectable phone language preference
  - AGI language helper, store default language for peer on creation
  - get-gs-language-by-ast-code dialplan helper - convert 2 letter asterisk language to gs/iso language
  - Changes to dialplan for i18n - may need further work
  - I18N enhancements for Polycom provisioning
  - Make user_add and user_change functions in inc/gs-fns/ language-aware
  - Missing bits for gs-users-get and gs-users-regen regarding i18n support
  - Updated english translation

gemeinschaft2.0 (2.3.1.3-lucid1) UNRELEASED; urgency=low

* New upstream revision

* LocaNet branch git/svn changes/additions since 2.3.1.2:
  - Anrufe auf den Parallelruf werden nun auch als angenommen/verpasst vermerkt.
  - Wurde von einer Warteschlange auf den Anrufbeantworter eines Users umgeleitet, so wurde dessen Voicemail-Ansage nicht abgespielt.
  - Es wurden im Queue-Monitor nicht alle, bzw. die falschen Queues angezeigt
  - Wenn man mittels Sudo ein anderer User wurde, konnte man nicht dessen Einleger fuer die Besetztlampen ausdrucken.
  - Ein Zeilenumbruch zuviel verursachte Fehlermeldungen auf der Manager-Schnittstelle und in der Asterisk-Konsole

gemeinschaft2.0 (2.3.1.2-lucid1) unstable; urgency=low

* New upstream revision

* LocaNet branch git/svn changes/additions since 2.3.1.1:
  - Faxe konnten nicht als PDF-Datei ueber die Web-Gui erzeugt werden.
  - Fehlerhafte Syntax fuer Kommentare in der Datenbankdefinition entfernt.
  - @ vor Funktion get_magic_quotes_runtime hinzugefuegt, da sonst AGI-Fehlermeldungen entstehen.
  - Asterisk Manager-Konfiguration fuer den AstButtond hinzugefuegt
  - Besetzt-Status beim Parallelruf mittels GROUP_COUNT() bestimmen
  - Datenbankdefinition korrigiert
  - Datenbank-Upgrade-Script von Version 2.0.2 auf 2.3.0
  - Umleitungen im Fall immer wurden immer als intern geroutet
  - Der User supervisor ist automatisch in der Gruppe admins.
  - Nur Admins (nicht normale User) sollten per Default die Berechtigung zum Umleiten von Warteschlangen haben.
  - Die Optionen -b/-w/-l/-d gibt es in sox nicht mehr in Ubuntu 10.04
  - Anpassungen fuer QueueMetrics 1.6.1 (Hotdesking)
  - Neue Version von qloader.pl
  - Anrufe, die auf die Voicemail-Box geleitet werden, werden wieder als verpasst gezaehlt.
  - Snom821-Support hinzugefuegt
  - Bei allen eingehenden Anrufen wurde der eigene Name als Anrufer-Name in Dial-Log gespeichert.
  - Labels fuer die Besetztlampenfelder von Snom und Grandstream-Telefonen werden automatisch erzeugt (als PDF).
  - Parallelruf und Zeitsteuerung als Funktionstasten am Snom-Endgeraet verfuegbar (mit Astbuttond)
  - agent-pause-unpause.agi aufgeraeumt
  - Agenten-Pause loggt nun auch PAUSEALL und UNPAUSEALL in queue_log
  - iaxmodem unterstuetzt keine call tokens
  - globale Hilfsvariable fuer QueueMetrics Auswertung von call outcomes eingebaut
  - Der AstButtond wird durch UserEvents ueber Aenderungen in Gemeinschaft unterrichtet und nichtmehr direkt ueber eine Socket-Verbindung.
  - Wartemusik in MeetMe-Konferenz abspielen, wenn Anrufer der einzige Teilnehmer ist

gemeinschaft2.0 (2.3.1.1-lucid2) UNRELEASED; urgency=low

* Add sudoers.d file
* Write version information to package tree again
  (introduced in 2.0.2.2-hardy1)
* Changelog entry for 2.3.1.0-hardy1 modified to include an overview
  of upstream changes

* LocaNet branch git/svn changes/additions since 2.3.1.0:
  - Penalties fuer Agenten aus der stable-Branch uebernommen.
  - Fehler in der Zuordnung der GUI-Module beseitigt
  - Erweiterung des Gruppen-Verwaltung aus Gemeinschaft hinzugefuegt.
  - Gui fuer die Verwaltung der angezeigten Gui-Module aus Gemeinschaft
    portiert
  - Es wurde an einigen Stellen in die Slave-Datenbank geschrieben.
    Patch aus der stable-Branch (f988491a099120f7f1ae8133584cd310287fdf9a)
    uebernommen.
  - Es konnte vorkommen, dass der Astbuttond die Verbindung auf die
    Socket-Schnittstelle verweigerte, da die Verbindung von der falschen
    Quell-Ip aus iniziiert wurde. Patch aus der stable-Branch
    (94e25a286e9da36dd24f30714157420dca303ac0).
  - Bei ausgehenden Gespraechen ueber ein Gateway wird der Inhalt dest
    Feldes dst in der cdr-Tablle nun in das Feld userfield geschrieben.

gemeinschaft2.0 (2.3.1.1-lucid1) unstable; urgency=low

* Import to lucid repository

gemeinschaft2.0 (2.3.1.1-hardy1) unstable; urgency=low

* New upstream version

gemeinschaft2.0 (2.3.1.0-hardy1) unstable; urgency=low

* New upstream version for Asterisk 1.6, synced to Gemeinschaft 2.3.1 codebase:
  - Fine-grained group permission system
  - Enhanced call diversion configuration including parallel calls, multiple
    voicemail announcements possible
  - AMI configuration interface
  - Improved CallerID manipulation options for gateway groups
  - Exportable private phone book

* Contains wakeup-call and room cleaning feature
* Patches for LocaPhone and path fixup updated
* Adjust conffiles

* LocaNet branch git/svn changes/additions:
  - Fixes for Asterisk 1.6 integration, fixes for new PHP versions, general
    code fixes and cleanups
  - Provisioning defaults adjusted

  From GIT log:
  - Feature-Codes fuer neue Umleitungsziele implementiert und definiert
    e-user-config.ael vereinfacht.
  - Die BLF-Taste am snom-Telefon wird nun korrekt provisioniert
  - Berechtigungsmodell repariert und erweitert. Rufumleitungen auf
    AB-Ansage repariert
  - Hilfe zu Feature-Codes aktualisiert
  - Berechtigung fuer die 'AB-Konfiguration' hinzugefuegt.

LocaPhone VoIP TK-System Release 2.0

gemeinschaft2.0 (2.0.2.1-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* LocaNet branch git/svn changes:
  - [Bug#226] Die Seite Rufannahmegruppen ist die Standardseite im Menuepunkt Monitor.
  - iPhone-Schnittstelle: Rufumleitungen auf die Ansage der Voicemail-Box sollten nun richtig angezeigt werden. Der Dummy fuer die externen freigegebenen Nummern wurde durch die tatsaechlichen freigegebenen Nummern ersetzt.
  - Als Ziel in einen IVR kann man nun auch eine Voicemailbox angeben.
  - Fehler im Datenbank-Update korrigiert
  - Bugfix: externen Anrufern bei ausgeschalteter Voicemailbox Besetzt statt 'Teilnehmer nicht erreichbar' signalisieren
  - [Bug#333] Unterstuetzung Endgeraet snom 820 hinzugefuegt.
  - [Bug#255] Das private Telefonbuch ist exportierbar. (Amooma-Patch)
  - Deletion of recordings requires user confirmation
  - [Bug#244] Anrufbeantworterabfrage immer mit PIN-Eingabe (konfigurierbar in der gemeinschaft.php)
  - gs-diallog-purge: Remove dial-log entries older than PROV_DIAL_LOG_LIFE from database to improve performance (to be run via cron)
  - Wenn ein angerufenes Telefon BUSY ist, dann muessen wir in user_uncallable_vm_off mit Hangup(17) statt Hangup() auflegen, sonst wird u.U. '480 Temporarily Unavailable' ueber SIP signalisiert
  - [Bug#430] Eine zweite Snom-Beistellung laesst sich nicht konfigurieren
  - [Bug#431] Fixed; caused by permission problems on the FIFO used e.g. for agi.log
  - [Bug#236] Sammelanschluss kann (optional) Besetzt signalisieren, wenn einer der Teilnehmer im Sammelanschluss besetzt ist. Funktioniert nur, wenn alle Mitglieder des Sammelanschlusses auf dem selben Asterisk-Server registriert sind
  - [Bug#432] An den Snom-Telefonen werden die verpassten Anrufe falsch dargestellt
  - More firmware revisions added to snom autoprovisioning, add missing field 'uniqueid' to ast_queue_members table (fixes crash bug with Asterisk 1.6.1.x)
  - Also set CALLERID(num) when faxserver places calls so that the user properly matches to the configured users and local dialplan rules can properly apply
  - konfigurierbarer Abwurf des Anrufes bei besetzt/nicht Melden/offline hinzugefuegt
  - User wurden beim Entfernen aus dem System nicht aus den Sammelanschluessen entfernt
  - [Bug#436] Er wurden Eintraege in das QueueLog geschrieben, obwohl der Anruf nie die Queue betreten hatte
  - DND-Taste auf dem Besetztlampenfeld von Snom-Telefonen in Verbindung mit dem AstButtond hinzugefuegt
  - Die Rueckwaertssuche aus dem oeffentlichen Telefonbuch wird nun auch fuer Warteschlangen durchgefuehrt
  - [Bug#437] Weitergeleitete verpasste Anrufe erzeugten keinen Eintrag in den Anruflisten
  - Das Aendern des Pause-Status von Agenten wurde nicht in das QueueLog geschrieben.
  - [Bug#71] Call-Diversion als Funktionstaste fuer die snom-Telefone hinzugefuegt
  - [Bug#438] Beim Entfernen eines Users aus dem System wurden nicht dessen Voicemails geloescht

gemeinschaft2.0 (2.0.2.0-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* Package changes:
  - "install" and "conffiles" files adjusted to latest git/svn changes
  - removed "druecken-sie-1...wav" as it is now contained in the git tree

* LocaNet branch git/svn changes:
  - Erste Version eines Proxies fuer den Gemeinschaft-Client auf dem iPhone
  - Verzeichnisstruktur fuer iPhone Proxy
  - Ist ein Benutzer an einem Telefon angemeldet, und er ruft die Webseite zur Konfiguration der Tastenbelegung auf, so wird zuerst das Profil fuer das Telefonmodell, an dem er angemeldet ist, angezeigt.
  - Custom-Kontexte aus opt/gemeinschaft/etc/asterisk/custom entfernt (wurden durch Kontexte im Verzeichnis /etc/gemeinschaft/asterisk ersetzt).
  - Datenbank 'asterisk' beim Upgrade benutzen
  - Auch die Funktion gs_user_phonemodel_get sollte den Usernamen und nicht die Extension des Users als Parameter erhalten
  - Satisfy Snom on SIP notify
  - gs-ast-ael-check ausfuehrbar gemacht
  - Optionaler SIP-Port wird jetzt nicht mehr ueber :Port im Hostnamen konfiguriert, sondern kann ueber die neu eingefuehrten erweiterten Parameter fuer jeden SIP-Node konfiguriert werden.
  - Polycom: Fix undefined variable PHP notice
  - Fehler in Agi-Script opt/gemeinschaft/dialplan-scripts/in-user-get-ringer.agi behoben. Es wurde immer nur der Default-Klingelton wiedergegeben
  - Siemens-Telefone unterstuetzen im Moment keine benutzerdefinierten Klingeltoene.
  - Es sind wieder zwei benutzerdefinierte Klingeltoene fuer snom-Telefone gleichzeitig moeglich

- Fehlerhaftes Pattern verhindert Einlegen der Rufumleitung auf die AB-Ansage ueber die Service-Nummer.
- Fehlender Sprachbaustein fuer das Sprachmenue hinzugefuegt.
- call-init-2.php funktioniert immer noch nicht
- ManagerEvents fuer neue Service-Nummern hinzugefuegt und das automatische Abmelden aus den Warteschlangen entfernt. Hilfe-Seite zu den Service-Nummern aktualisiert.
- An dieser Stelle sind keine Benachrichtigungen des Astbuttond ueber die Socket-Schnittstelle mehr noetig. Dieses geschieht jetzt ueber ein ManagerEvent
- Bugfix: In Anruflisten in der Web-GUI werden Vorname und Nachname nicht mehr durch Komma getrennt.
- Namensanzeige in Anrufliste von 'Vorname Nachname' in 'Nachname, Vorname' geaendert, um der Reihenfolge in den Telefonbuechern zu entsprechen.
- [Bug#377] Rufumleitungen direkt auf eine Mailbox sind nun auch fuer die Sammelanschluessen und die Warteschlangen moeglich
- Absendername und Email-Adresse fuer die Voicemail-Benachrichtigung konfigurierbar. MailHost-Name wird aus /etc/mailname gelesen, so vorhanden.
- Es wird nun ueberprueft, ob die Sekretaerin den Chef anrufen darf
- Individuelle Konfigurierung verschiedener Lautstaerke-Einstellungen des Endgeraetes ueber die Web-Gui per Auto-Provisionierung hinzugefuegt
- Fehlender HTML-Tag ergaenzt. Auf der Web-Seite fuer die Sammelanschluesse waren zwei Anfuehrungszeichen.
- Bei den Lautstaerkeeinstellungen werden nun auch die Aastra-Telefone unterstuetzt. Vorprovisionierte Lautstaerkeeinstellungen durch den Administrator werden als Defaulteinstellungen angezeigt.
- [Bug#360] Warteschlangen konnten nicht geloescht werden
- Neue Service-Nummern fuer die Rufumleitung (Ab/Ansage), Erweiterung der Hilfe-Seite und der Tastenbelegung.
- [Bug#265] Wird ein Gespraech umgeleitet, wird dem Angerufenen (intern) innerhalb des Callerid-Namen angezeigt, von wo das Gespraech umgeleitet wurde
- [Bug#321] Layout der Benutzerverwaltung fuer den Administrator ueberarbeitet
- Bugfix und Debug-Ausgabe fuer dialplan-scripts/check_secretary_permission.agi
- [Bug#385] Regulaere Ausdruecke in Waehlbefehl wurden nicht richtig ersetzt
- [Bug#352] Anzeige Zielrufnummer am Endgeraet bei Wahl aus WebGUI
- snom Autoprovisionierung: cancel_on_hold von 'on' auf 'off' geaendert, damit Gespraeche mit der PhoneSuite CTI gehalten und zurueck geholt werden koennen.
- Beispiel-Eintrag fuer die Sip-Gateways in der Datenbankdefinition war fehlerhaft
- [Bug#392] War ein Telefon im Sammelanschluss abgemeldet, so war der Sammelanschluss nicht mehr erreichbar
- Sprechgarnitur fuer einheitliche Bezeichnung mit dem Handbuch in Headset umbenannt. Fehlende Bilder hinzugefuegt.
- Die Tabellen auf den Hilfe-Seiten wurden im IE nicht richtig angezeigt
- Polycom: readd MESSAGE_WAITING tone and disable playback 'the Polycom way' to prevent phone crashed
- Der Sip-Header Alert-Info muss bei weitergeleiteten Anrufen entfernt werden, damit der benutzerdefinierte Klingelton richtig abgespielt wird
- Fehlender SIP-Header "forwarded_by" fuer den Clusterbetrieb ergaenzt
- When ring_instead_of_moh is not set, properly Answer() channels (apparently asterisk should do this...) to avoid trouble with certain SIP providers
- Phonebook reverse lookup: Remove check on existence of CALLERID(name): Phonebook entries should always override ANY name information
- Ist das Ziel bei einer Weiterleitung ohne vorherige Ankuendigung nicht erreichbar, so kommt das Gespraech zu dem Anschluss zurueck, von dem aus das Gespraech vermittelt wurde. Ist dieser Anschluss auch nicht erreichbar, so kann ein Standardziel definiert werden. Diese Funktion ist konfigurierbar.
- Da der Callerid-Name nun immer von den Telefonbuch-Eintraegen ueberschrieben wird, muss die Information, dass es sich um einen Rueckruf nach einer Weiterleitung ohne vorherige Ankuendigung handelt, nach der Suche im Telefonbuch vorgenommen werden.

* Changes imported from Amooma SVN:
  - Lots of code cleanup and improved logging, AGI scripts etc.
  - Updated & improved online help in many parts/sections, updated handling of Docbook->XHTML conversion
  - Improvements to configuration and configuration handling
  - Updates to database definition
  - Grandstream support improved, pickup added, custom ringtones, phonebooks
  - Additions and fixes to Phonebooks and dial logs
  - Outbound SIP proxy support
  - All SIP peer parameters can now be changed

gemeinschaft2.0 (2.0.1.5-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* LocaNet branch git/svn changes:
  - Move default location of astbuttond displayfiles directory to /var/spool/astbuttond

gemeinschaft2.0 (2.0.1.4-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* LocaNet branch git/svn changes:
  - Polycom: idlescreen display (missed calls, dnd, queue states etc.); requires current/updated astbuttond which writes the idle display info into the filesystem to work.
  - XML Push-Benachrichtigung fuer Polycom Endgeraete entfernt, weil sie manche Geraetemodelle zum Absturz bringt.
  - Fehlender Befehl zum Verwerfen der Tabelle ivrs in der Datenbank-Definition hinzugefuegt
  - Damit beim Update der Datenbank nicht die falschen Tabellen geloescht werden, wurde das Datenbank-Update von Version 1 auf Version 2 ausgegliedert
  - Update fuer PAMAL: LDAP-Benutzer koennen jetzt auch autorisiert werden, wenn sie sich in einer OU unterhalb von LDAP_SEARCHBASE befinden.
  - Voicemails koennen optional nach dem Versand per Email auch automatisch geloescht werden.

gemeinschaft2.0 (2.0.1.3-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* LocaNet branch git/svn changes:
  - Die Channel-Variable is_from_gateway wurde im Cluster-Betrieb nicht aus dem Sip-Header ausgelesen
  - Wurde ueber die Service-Nummer am Telefon eine temporaere Rufumleitung eingelegt, so wurde der Timeout bei keine Antwort auf 20 Sekunden gesetzt
  - Snom: Add language pack detection for firmware 7.3.23

gemeinschaft2.0 (2.0.1.2-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* LocaNet branch git/svn changes:
  - Siemens: Add create table sql file for prov_siemens table to documentation folder

gemeinschaft2.0 (2.0.1.1-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)

* Package changes:
  - Flag /opt/gemeinschaft-siemens/conf.php as configuration file
  - Rename idlescreen.png and logoscreen.png to idlescreen.os80.png and logoscreen.os80.png to match fixes for phonetype-based filenames, and also add copies for os60 series and a monochrome version for os40
  - Add polycom logs dir to dirs file and fix owner for logs dir in postinst

* LocaNet branch git/svn changes:
  - Siemens: (re)Implement %s-replacement for idlescreen and logoscreen configuration vars
  - Siemens: Updated prov-settings.php and corresponding conf.php from upstream

gemeinschaft2.0 (2.0.1.0-1) unstable; urgency=low

* Package changes:
  - Add idlescreen.png and logoscreen.png for siemens provisioning
  - Remove firmware package dependencies but rather recommend them
  - Flag e-ivr.ael.php as conffile

* LocaNet branch git/svn changes:
  - Patch fuer IVR-Funktion hinzugefuegt und ueberarbeitet
  - Fehlendes Include hinzugefuegt. Anrufe auf die Voicemail-Box werden als verpasst bewertet.
  - Bug#364: We don't want the phone to interpret our digits at all...
  - In den Anruflisten wird neben der Rufnummer auch der Name des Anrufers bzw. des Angerufenen angezeigt, so die Nummer im persoenlichen oder im Firmen-Telefonbuch hinterlegt ist
  - Bug#359: Do full select on all available recordings and use the result to populate the playback dropdown item and also get the number of total entries from that select so that the forward/back buttons work.
  - Polycom: Map Gemeinschaft GUI callwaiting option to reg.1.callsPerLineKey
  - Polycom: Add support for Polycom phones PUTting their logfiles onto the webserver (configuration tunable)
  - snom Firmwareupdate funktioniert wieder (Danke an Chris Halls)
  - Benutzername bei der Anmeldung an der Weboberflaeche in Kleinbuchstaben umwandeln, da Gemeinschaft intern nur Kleinbuchstaben erlaubt.
  - Remove agent from queues on delete

    Previously, the delete button would not work if the agent was a member of any queue, due to a database constraint. There was no indication this was the problem, except for an SQL error in the database log.
  - Siemens OpenStage Provisionierung
  - Dokumentation fuer Siemens Autoprovisionierung

gemeinschaft2.0 (2.0.0.11-hardy1) unstable; urgency=low

* Package for Ubuntu "Hardy" Server (8.04 LTS)
* Depend on ncurses-term to fix dialplan generation problems
  (ncurses-term provides proper "unknown" terminfo file)

gemeinschaft2.0 (2.0.0.11-1) unstable; urgency=low

* Package changes:
  - gemeinschaft2.0-moh is no longer a dependency of gemeinschaft2.0

* LocaNet branch git/svn changes:
  - Eingabefeld fuer Benutzername auf Login-Seite auf 50 Zeichen verlaengert
  - Routing/BRI: Make Dialstring editable
  - Unterstuetzung fuer den Gruppenpickup hinzugefuegt
  - Polycom-Telefone sollen auch die Raute waehlen koennen
  - Da ein Snom-Telefon aus dem Telefonbuch nur Nummern ohne Buchstaben oder andere Sonderzeichen waehlen kann, werden solche Zeichen vor dem Waehlen entfernt.
  - Polycom: Fix ringback tone delay
  - Bugfix: Beim Agentenlogin und Agentenlogout wurde das SIP-Peer nicht mit in queue_log geschrieben
  - Da alle Benutzer in der Pickupgruppe 1 waren, konnte jeder Benutzer mit *8 ein Gespraech picken. Benutzer sind von nun an in keiner Pickupgruppe mehr.
  - Die Auswahl der Callerids ueber den Minibrowser der Snom-Telefone ist wieder moeglich
  - Bugfix: beim Abmelden aus einer Warteschlange wurde nichts in queue_log geloggt
  - Optimierung des Queue-Loggings, so dass nur noch Asterisk-interne Funktionen und nicht mehr System() benutzt werden.
  - Bugfix: aus Telefonbuch ausgeblendete Benutzer nicht anzeigen
  - [Bug#362] Anrufe, die auf den Anrufbeantworter weitergeleitet werden, werden als verpasst gezaehlt.
  - Die Ansage fuer nicht erreichbar kann vom Benutzer geloescht werden
  - Ansagen werden nur gespeichert, wenn die Aufnahme mit der #-Taste beendet wurde
  - Der Pause-Status fuer Agenten wird nun in der Realtime-Datenbank gesetzt
  - Um zu verhindern, dass der Reload-Knopf zweimal hintereinander gedrueckt wird, wird er durch ein JavaScript gesperrt
  - Wird ueber die Weboberflaeche gewaehlt, und die angerufene Person war besetzt, so wurde dies nicht signalisiert
  - Polycom: Remove softkey remappings to fix Back/Exit navigation in the microbrowser

* Changes imported from Amooma SVN:
  - OpenStage: Tastenfunktion 14 (deflect): s/Abweisen/Weiterleiten/
  - Hinweis bei MSIE <= 6
  - GRANDSTREAM_PROV_HTTP_PASS auf Default "admin" gesetzt.
  - Fix: Verpasste anrufe ohne Namen, Patch von Sascha Daniels
  - Beispiel-User-Agent-HTTP-Header des Aastra 53i als Kommentar hinzugefuegt
  - Grandstream GXP 280 support

gemeinschaft2.0 (2.0.0.10-1) unstable; urgency=low

* Cleanup work on Polycom APS code, support for SoundStation IP conference phones
* General code cleanup and typo fixing
* Autoprovisioning for Elmeg IP290 SIP phones, based on Snom autoprovisioning
* Caller ID Name reverse lookup
* Configurable option to allow or deny of calling external targets for nobody users
* Workarounds for public SIP providers other than sipgate
* Snom phone menus (callforward configuration) updated
* Bugfix for queue/huntgroup redirect GUI
* Updated and fixed queue announcement options
* gs-cc-worker no longer triggers for DND-enabled users
* Subqueue support for queuemetrics
* RC script for qloader.pl in documentation
* Usernames may now contain .-char
* Statistics are now only visible for the admin

gemeinschaft2.0 (2.0.0.9-1) unstable; urgency=low

* Removed provisioning of XHTML application references for Polycom IP500
* Support for custom context in the default context
* call-init-2.php reverted to an earlier revision
* Polycom IP300 and IP500 SIP application reference changed to 2.1.2
* Polycom directory.php code cleanup

gemeinschaft2.0 (2.0.0.8-1) unstable; urgency=low

* Bugfix for syntax error in sip-nodes.conf.php
* Show missed calls on Polycom phones
* Added LDAP support for web authentication
* Added support for multiple web authentication methods

gemeinschaft2.0 (2.0.0.7-hardy2) unstable; urgency=low

* gemeinschaft2.0-core now depends on libsox-fmt-all

gemeinschaft2.0 (2.0.0.7-1) unstable; urgency=low

* Fixed defaults.xml

gemeinschaft2.0 (2.0.0.6-1) unstable; urgency=low

* Polycom provisioning updated to support legacy SoundPoint phones (ip300, ip500)

gemeinschaft2.0 (2.0.0.5-1) unstable; urgency=low

* Autoprovisioning for Polycom SoundPoint phones

gemeinschaft2.0 (2.0.0.4-1) unstable; urgency=low

* Import of queue_log can now be turned off
* Disabled unsupported ringtones on snom phones
* optionally ask for destination number on call forward (*2#)
* optionally do not log missed queue calls

gemeinschaft2.0 (2.0.0.3-1) unstable; urgency=low

* New upstream revision

gemeinschaft2.0 (2.0.0.2-1) unstable; urgency=low

* New upstream revision (all upstream LocaPhone traces finally removed)

gemeinschaft2.0 (2.0.0.0-1) unstable; urgency=low

* Upstream is now based on GIT, all patches except LocaPhone are already imported

gemeinschaft2.0 (0.0.0+svn6321-1) unstable; urgency=low

* New upstream revision
* New patches 126-clipexternfix.diff, 127-clipnull.diff, 128-gwdialstring.diff and 129-snomfirmware7139.diff
* Added quilt to build dependencies

gemeinschaft2.0 (0.0.0+svn6308-1) unstable; urgency=low

* New upstream revision

gemeinschaft2.0 (0.0.0+svn6304-4) unstable; urgency=low

* New patches 123-calleridfix.diff, 124-snomprovisionrefresh.diff and 125-snomanruflisteloeschen.diff

gemeinschaft2.0 (0.0.0+svn6304-3) unstable; urgency=low

* Updated 118-snomsettings.diff
* New patches 121-snomheadset.diff, 122-asterisksqldefaults.diff and aastradefaults.diff

gemeinschaft2.0 (0.0.0+svn6304-2) unstable; urgency=low

* Additional patches (115-klingeltoene.diff, 116-pickupsammelanschluss.diff, 117-sipreinvitenode.diff, 118-snomsettings.diff, 119-dbupgradefix.diff, 120-asterisksqlfix.diff)

gemeinschaft2.0 (0.0.0+svn6304-1) unstable; urgency=low

* Initial package - LocaPhone 2.0