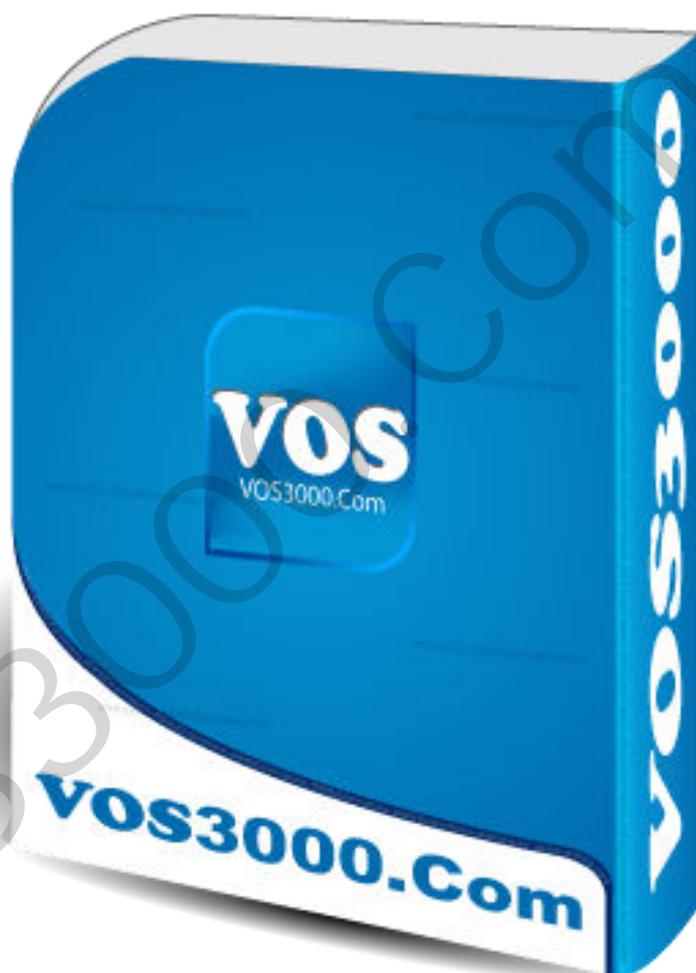


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About This Document

Purpose

This document describes the functions and operations of VOS3000 client.

Intended Audience

This document is intended for:

- Installation and commissioning engineers
- NM configuration engineers

Symbol Conventions

The symbols that may be found in this document are defined as follows.

Symbol	Description
 DANGER	Indicates a hazard with a high level or medium level of risk which, if not avoided, could result in service interruption.
 WARNING	Indicates a hazard with a low level of risk which, if not avoided, could result in data missing.
 CAUTION	Indicates a potentially hazardous situation that, if not avoided, could result in unanticipated results.
 TIP	Provides a tip that may help you solve a problem or save time.
 NOTE	Provides additional information to emphasize or supplement important points in the main text.

Change History

Changes between document issues are cumulative. The latest document issue contains all the changes made in earlier issues.

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1 Guide

About This Chapter

This chapter describes the user guide of VOS3000 client.

Operation Guide

In VOS3000, most data managements can be completed through sheets. Sheets can be opened by double-clicking corresponding nodes in “Navigation”. The following operations are supported:

- Open: open current management page.
- Filter: get current configuration from server.
- Copy: copy the currently selected sheet line into the clipboard.
- Paste: paste the line in the clipboard into a sheet with the same type.
- Add: insert new lines.
- Delete: delete sheet lines. If the data are at the server, the selected lines will be marked as “to be deleted”.
- Apply: send currently specified operations (such as add, delete and modify) to the server to carry out.



NOTE

Before clicking “apply”, all the operations of data are saved only at the client end and will not affect the server's data; closing the management page would discard these operations.

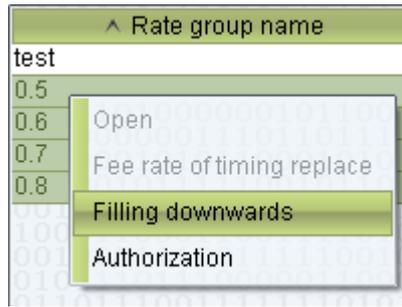
- Export: export the current sheet into local files.
- Import: import data from local files into the sheet (supported by a few types of sheets).



NOTE

Batch data operations can be fulfilled by “copy”, “paste” and column “Filling downwards” functions supported by spreadsheets in VOS3000. See the figure below:

Figure 1-1 Filling downwards



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2 Function

About This Chapter

This chapter describes the functions of VOS3000.

2.1 Login

Upon running VOS3000 client, the login dialogue will be shown.



- Server ip: IP address and login port of the remote server.
- User name: user names allowed by the platform.
- Password: user password allowed by the platform.
- Uuid: during the first installation, system will generate this id.

Even password is correct, uuid is still needed. This id can be modified any time, please refer to <Configuration Guide>.

The system will record IPs typed by users for later use. Users can also delete these historical servers IP.

The initial user name and password are admin and admin.



CAUTION

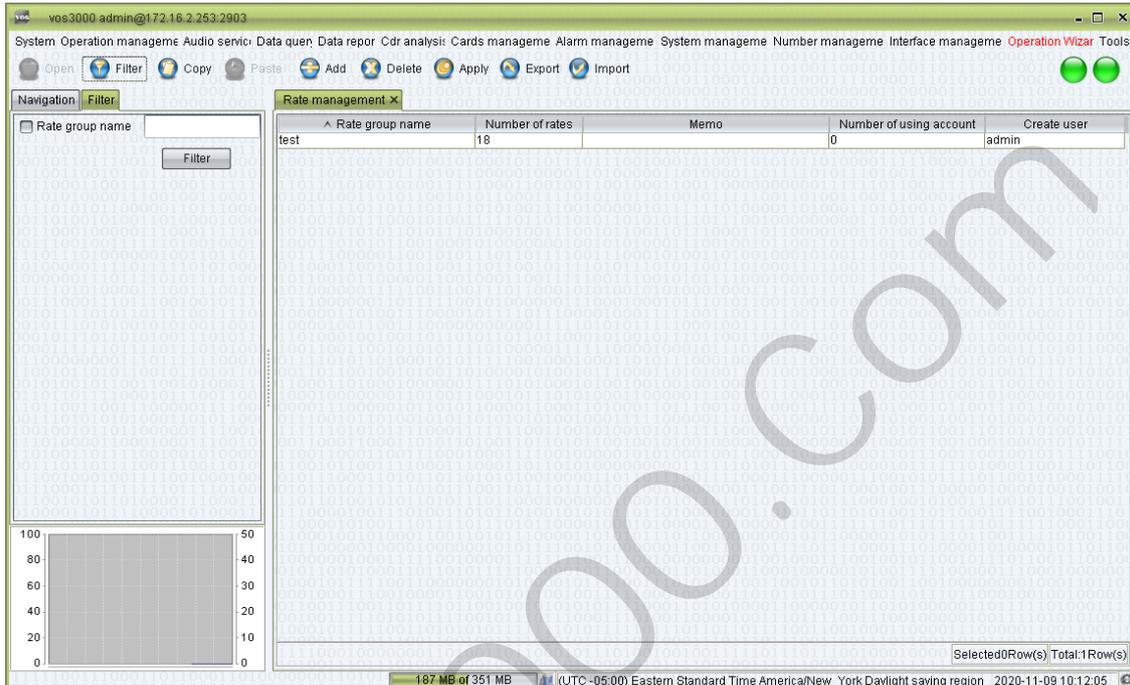
Please modify the initial login password as soon as possible.

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2.2 Rate Management

2.2.1 Rate Group Management

This function is used to define rate groups used for billing.



How to Start

- Double-click “Navigation > Rate management”

Table Items

- Rate group name: the name of the rate group. When new accounts are created, one rate group must be specified and the name here will be used to identify the groups.

NOTE

Try to choose more informative names that remind people of the rate's details.

- Number of rates: the number of rates contained in the group.
- Equivalent time to 60 seconds for calling card: for calling card only.
- Memo: additional comments.
- Number of using account: the number of accounts using this rate group.
- Create user: the name of the user who created this rate group.

Other Operations

- Double-click the numbers at “Number of rates” to enter the rate management page.
- Double-click the numbers at “Number of using account” to enter the account management page.

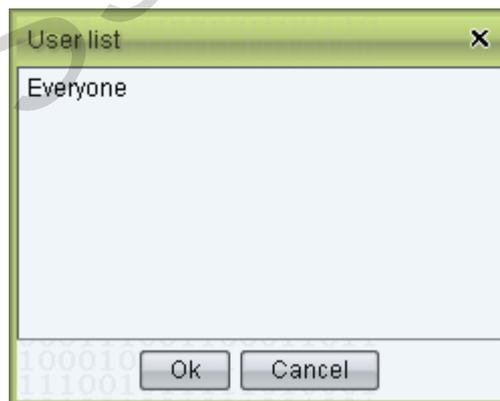
Right-Click Menu



- Open: enter the rate management page.
- Fee rate of timing replace: add timing replace fee rates.
- Filling downwards: use for copy information.
- Authorization: enter the authorization management page.

 **NOTE**

You can simultaneously select multiple rate groups.

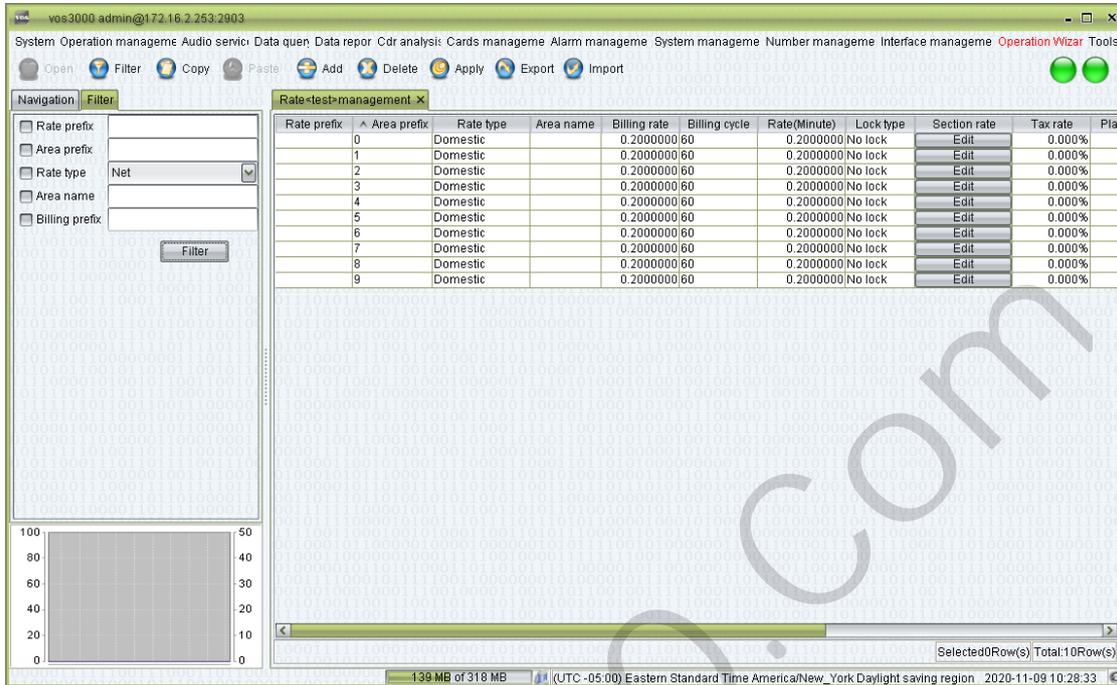


 **NOTE**

For agent, when login system, only those rate groups in the “Authorization” lists will be shown. As illustrated above, this rate group can be seen by “agent1”, “agent2”and “agent3”.

2.2.2 Rate Management

This function defines the prefixes to each number during the billing.



How to Start

- Select a line in the rate group management page and choose “Open” in the right-click menu.
- Double-click the “Number of rates” in the rate group management page.

Filter

- Billing prefix: input billing number to find used billing rate.
- Add domestic area code(full): add Chinese mobile phone prefix 013,015,018 and all area code defined by “City code” automatically.
- Add domestic area code(brief): add prefix 01-09 automatically.

Table Items

- Rate prefix: the prefix of the called number which matches this rate.



NOTE

The longest matching prefix will be used. For example, if there are two rate prefix, “0” and “01”, the number “01117” will be match to “01” since it is the longest pattern that matches “01117”.

- Area prefix: also used for rate matching, area name will be shown according this, please refer to “Number management > Area information”.
- Rate type: available options are “Net”, “Local”, “Domestic” and “International”.



NOTE

The type will not only be shown in CDR, but also used in many filters and statistics. Please correctly specify the rate type. Meanwhile, this type will be checked before calling. If a caller does not have the authorization to call the type of number specified here, the call will be banned.

- Area name: the area corresponding to the rate prefix. Non-editable, which is set by “Number management > Area information”, displays the “Area information” of “Area prefix”.
- Billing rate: the fee charged for each billing cycle.
- Billing cycle: the session time that exceeds the “First time duration” will be divided into units with length specified here. When the time cannot be divided clear, it will be rounded up to the nearest integer.

NOTE

An example: if the “First time rate” is “0.21”, the “First time duration” is “180”, the “Billing rate” is “0.15”, and the “Billing cycle” is 60, then according to this rate, a session that lasts 250s will be charged $0.21 + 0.15 * 2 = 0.51$.

- Rate(Minute): calculate by “Billing rate” and “Billing cycle”.
- Lock type: enable or disable the rate.
- Section rate:

Serial number	Money amount	Charged duration
0	0.5000000	60
1	0.4000000	60
2	0.3000000	60

- Serial number: time section.
- Money amount: money of this section.
- Charged duration: charge time.

NOTE

Section rate will be used first, sessions that exceeds this time will be charged according to the “Billing rate” and “Billing cycle”. If none section is set, the “Billing rate” and “Billing cycle” will be used from the beginning. (Unit: Second).

- Billing rate for calling card prompt: used for IVR prompt remaining duration.
- Billing cycle for calling card prompt: used for IVR prompt remaining duration.
- Tax rate: used for daily tax increases.cdr not show tax rate cost,account and report show tax rate+base rate.

Other Operations

- The table supports “Import” and “Export” operations.

NOTE

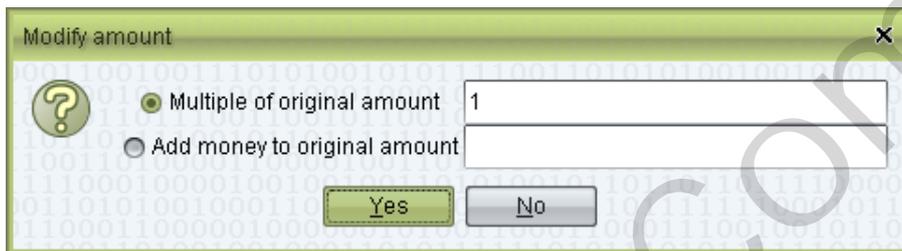
Refer to the exported files for the import format.

Right-Click Menu

- Modify amount: on column “Billing rate”.



- **Modify amount:** select the need to modify the amount of the line, when the mouse is in the <Billing rate> or <Rate(Minute)>, can be selected on the amount of line adjustment.



- **Multiple of original amount:** the original <Billing rate> amount multiplied by the set value.
- **Add money to original amount:** the original <Billing rate> amount add the set value.
- **Split area prefix:** for new line only.

Rate prefix	^ Area prefix	Rate type
	1,2,3,4,5	Domestic

Rate prefix	^ Area prefix
	1
	2
	3
	4
	5

- **Combine rate prefix:** combine “Rate prefix” and “Area prefix” to “Area prefix”.

Rate prefix	^ Area prefix	Rate type
88	1	

Rate prefix	^ Area prefix
	881

- **Add start bit of prefix:** when selected lines are in the status of “to be added”, the function can be used to modify their initials.

- Delete start bit of prefix: when selected lines are in the status of “to be added”, the function can be used to modify their initials.
- Fee rate of timing replace: At a certain time, modify the setting rates.

Shortcut	Time of timing replace	Mode of timing replace	Rate prefix	^ Area prefix	Rate type	Area name	Billing rate
Fee rate of timing replace<demo>management x	2018-12-07 00:00	Append replace		2	Domestic		0.0500000
	2018-12-07 00:00	Delete		3	Domestic		0.0000000



NOTE

Some international rate list use nonstandard initials. These functions can be used to adjust the initials after import.

When creating rate for national calls, try to use rate prefixes like “01” to “09” instead of using a single “0”. Otherwise, if the international rate are incomplete, some international calls might be matched to “0” and misclassified as national calls.



TIP

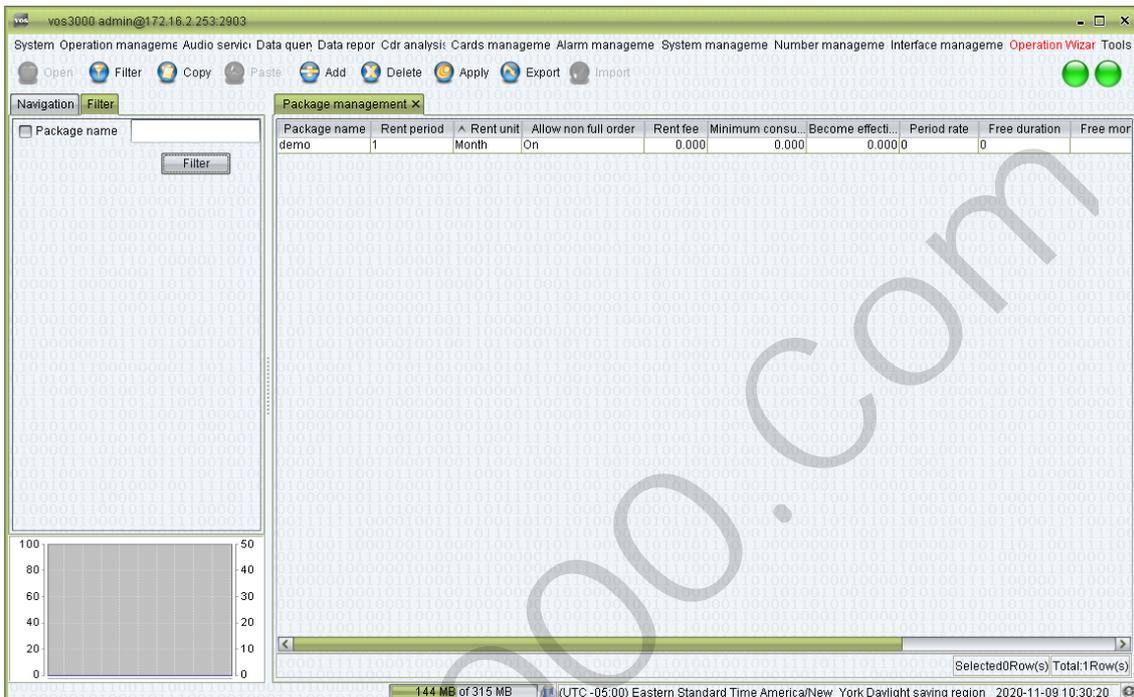
Some provider gives minute cost only, when import, you can set “Rate(Minute)”, system will calculate “Billing rate” automatically. You can use split area prefix and combine rate prefix to modify rates, obtain the rate which can be vos system accepted.

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2.3 Package Management

2.3.1 Package Group Management

This function is used to define packages for billing.



How to Start

- Double-click “Navigation > Package management”

Table Items

- Package name: the name of the package.
- Rent period: value of time.
- Rent unit: Day/Month/Year.

NOTE

Package’s period = Rent period * Rent unit, e.g. rent period: 7, rent unit: day, means 7 days package, every 7 days will effect until expire.

- Allow non full order:
 - On: if package order is not on the first day of subscription period, the rent is calculated from ordered day to expired day of package.
 - Off: if package order is not on the first day of subscription period ,then tent full deduction,The rent will be fully deducted.

- Rent fee: Package cost.

NOTE

- If rent fee is 0, rent will be always subscribed.

- Minimum consumption: calculate when package expired, if account under consumption, the difference will be deducted.
- Become effective spending limit: only if period suite consumption more than or equal to this value ,this package can be used.
- Period rate: the rate of a certain time period. Double-click to open the period rate management. Please refer to the next section for details.
- Free duration: free sessions provided by the package. Double-click to edit it.

Package<demo>free duration management x				
^ Begin time	End time	Area prefix	Free duration	Billing cycle
00:00:00	24:00:00	0:	0	6

- Begin time: start time of gift duration
- End time: end time of gift duration
- Area prefix: billing number prefix
- Free duration: seconds given away
- Billing cycle: billing period corresponding to the free time
- Free money amount: gift money provided by the package. Double-click to edit it.



NOTE

The free duration will be used first, then the free money amount.

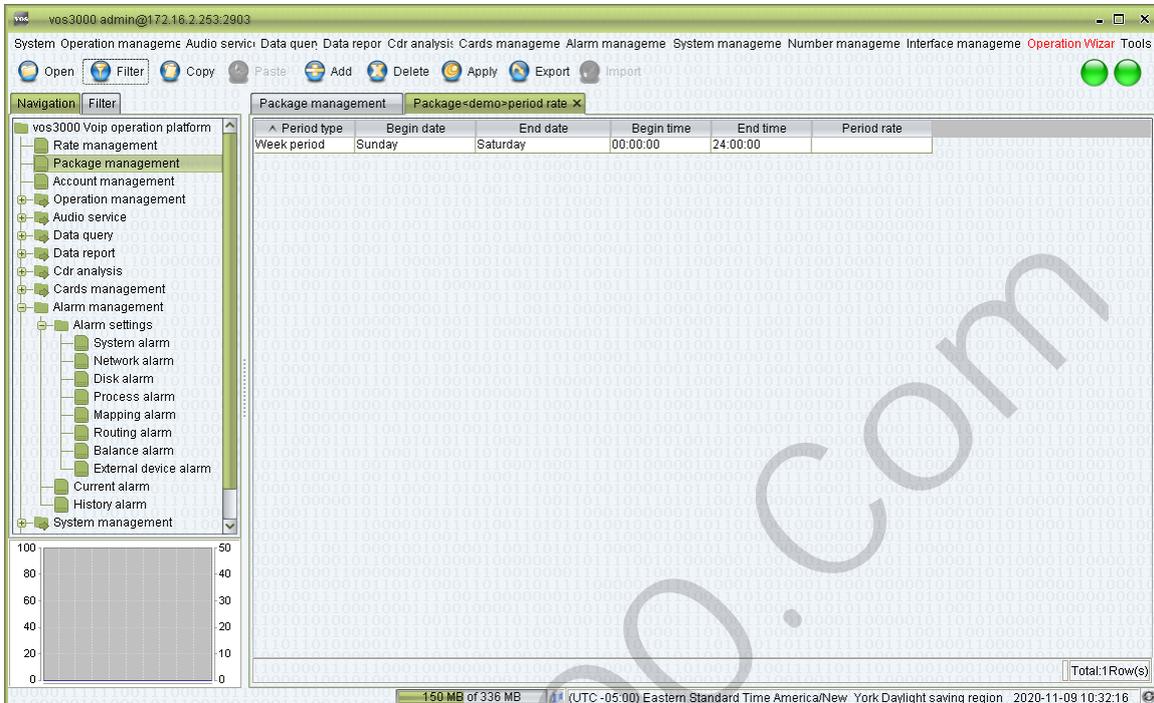
- Memo: additional comments.
- Create user: the name of the user who created this package.

Right-Click Menu

- Authorization: refer to rate management.

2.3.2 Package Period Rate Management

This function is used to define packages for period rate, one package can define different periods with different billing rates.



How to Start

- Double-click the numbers at the “Package management > Period rate”

Examples:

- The rate in the weekends is “0.5”.

^ Period type	Begin date	End date	Begin time	End time	Period rate
Week period	Sunday	Saturday	00:00:00	24:00:00	0.5

- The rate from 0:00 to 8:00 is “0.5” and “0.6” for the rest of the time.

^ Period type	Begin date	End date	Begin time	End time	Period rate
Week period	Sunday	Saturday	00:00:00	08:00:00	0.5
Week period	Sunday	Saturday	08:00:00	24:00:00	0.6

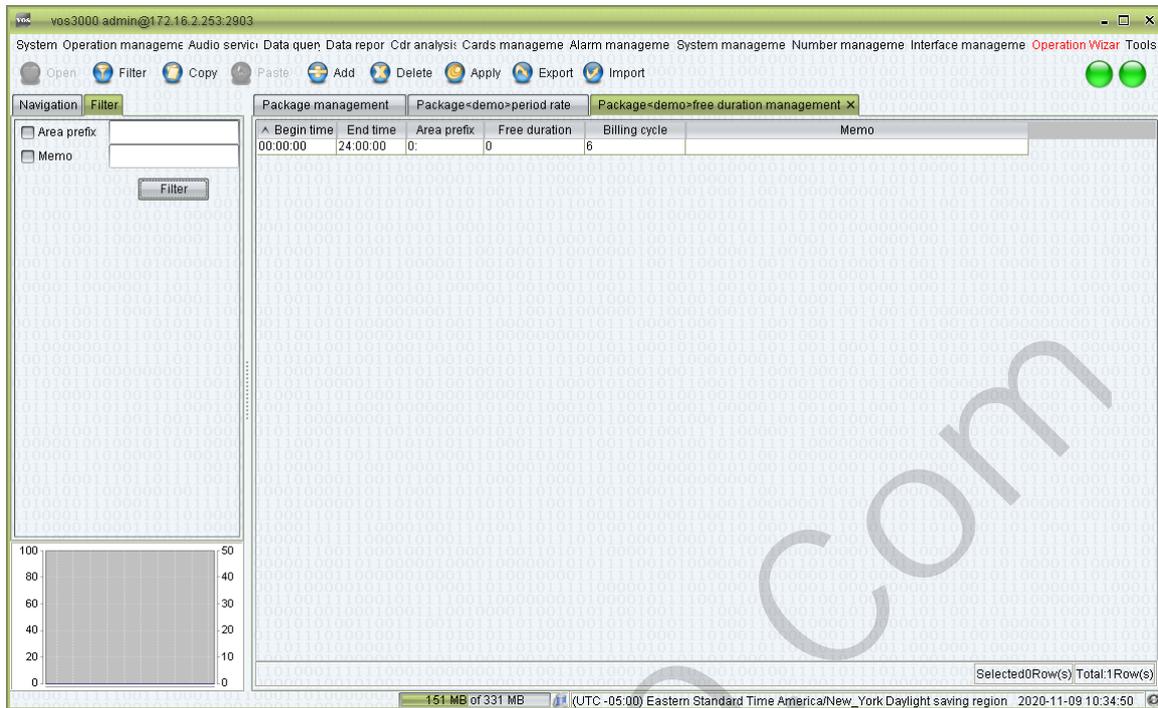
- The rate from 0:00:00, December 6th, 2018 to 20:30:00, January 30th, 2019 is “0.5”.

Period type	^ Begin date	End date	Begin time	End time	Period rate
Year period	2018-12-06	2019-01-30	00:00:00	20:30:00	0.5

NOTE

The start and expiration time for yearly periods are different from those for weekly or monthly ones.

2.3.3 Package Free Duration Management



How to Start

- Double-click “Navigation > Package management > Free duration”

Table Items

- Begin time: the begin time of free duration.
- End time: the end time of free duration.



NOTE

Call time within Begin Time and End Time, free duration will be used. If not whole day time, free duration is from call start to End Time.

- Area prefix: the prefix of free duration.
- Free duration: free time for the prefix.
- Billing cycle: charge cycle.
- Memo: comments on the package.

Right-Click Menu

- Authorization



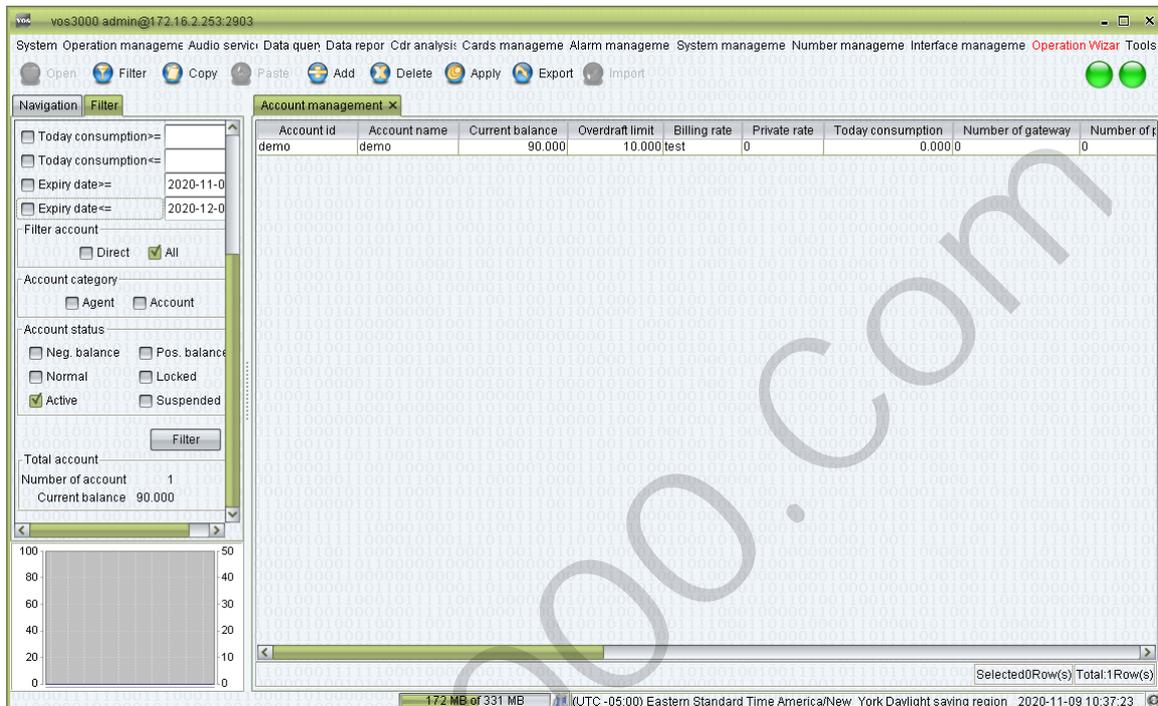
NOTE

Please refer to: rate group management.

2.4 Account Management

2.4.1 General Account

This function is used to manage all billing accounts of the platform.



How to Start

- Double-click “Navigation > Account management”

Table Items

- Account id: the unique identification of the account. This must be unique and cannot be modified once the account is created.
- Account name: the name of the account (such as the full name of the user).
- Current balance: the current balance of the account.
- Overdraft limit: the maximum credit of the account.
- Billing rate: the rate group for billing.
- Private rate: rate for this account only, if billing rates contains private rate, the lower will be chosen.
- Today consumption: today's consumption.
- Number of gateway: non-editable. The number of gateways under this account. Double-click to enter the gateway management page of this account.
- Number of phone: non-editable. The number of phones under this account. Double-click to enter the phone numbers management page of this account.
- Suite order: non-editable. It shows the number of packages subscribed by this account.
- Current suite: non-editable. The number of packages effected.

- Agent id: the “Account id” of its parent account. The parent account must exist. Upon designation, the parent account will become the “Agent” type.
- Additional settings: information about the customer. Click “Edit” to change.
- Directory: non-editable. The number of this account’s phonebook. Double click to enter short number settings.
- Directory limit: number of phonebooks.
- Account category: “Account” or “Agent”, non-editable. When an account has sub accounts, it automatically becomes an agent.
- Account type: “General”, ”Phone card” or “Clearing”.
- Memo: comments on the account.
- Account status: “Normal” or “Locked”.
- CTD billing model: just used for callback business, and account is “Agent” type.
 - Standard: callback billing mode of “Agent” account is decided by the subsidiary account’s in callback business.
 - Flow: callback billing mode of “Agent” account is decided by its own setting in callback business.

 **TIP**

CTD billing mode setting is often used for the condition where the first line free billing and the second line billing in callback business.

Expiry date: the expiration date of the account.

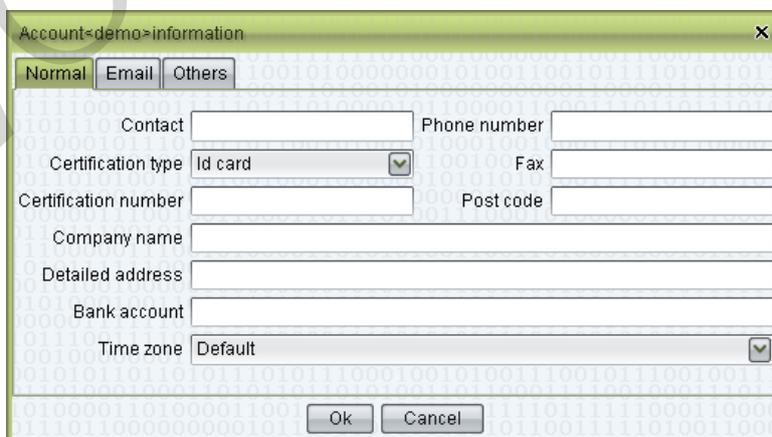
- Advance amount: advanced money for ongoing calls.

 **TIP**

Need enable anti overdraft

- Advance amount: advanced money for ongoing calls.
- Expiry date: validity period of the account
- Date of open account: start time of account creation, can’t edit
- Additional settings

Additional settings > Normal



- Record the basic information of the account

Additional settings > Email

- Fill in the basic information of push report to the third party server

Additional settings > Others

- Pickup access code end of the account:

when the phone on the vos3000 is ringing, other phones can use the proxy access code to receive the call of the phone. The only principle of whether the account terminated by the two phones is the same.

if “Pickup access code end of the account” is checked in the direct account of the phone, only other phones under the direct account can be substituted. If this item is not checked, the agent parent account of the phone is in a favorable situation and the “Pickup access code end of the account” option of the agent parent account is queried. If it is still not checked, the check will continue.

when the final account has no agent parent account, it is considered to be terminated in empty. If another phone ends up with the same account as this phone, it can take the call from this phone.

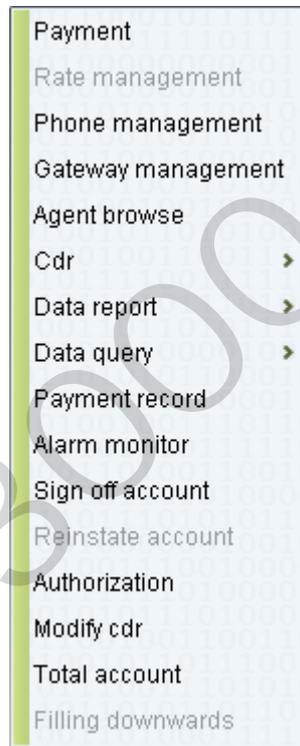
- Enable anti overdraft: prevent the call from exceeding the preset amount, and use it with the advance amount
- Suppressing all duration too long alarm: abnormal call alarm during working hours
- Alarm email: alert recipient email

- Work calender: set the working time and non working time to suppress the call volume alarm

Other Operations

- Double-click the number at “Number of gateway” to enter “Mapping gateway management” page for the account.
- Double-click the number at “Number of phone” to enter “Phone number management” page.
- Double-click the number at “Number of package” to manage the packages subscribed by the account.
- Double-click the number at “Current package” to edit the effected package of the account.

Right-Click Menu



- Disable account: disable the account and all its sub accounts. Phone numbers belonging to these accounts will not be able to make phone calls.
- Enable account: enable the disabled account.



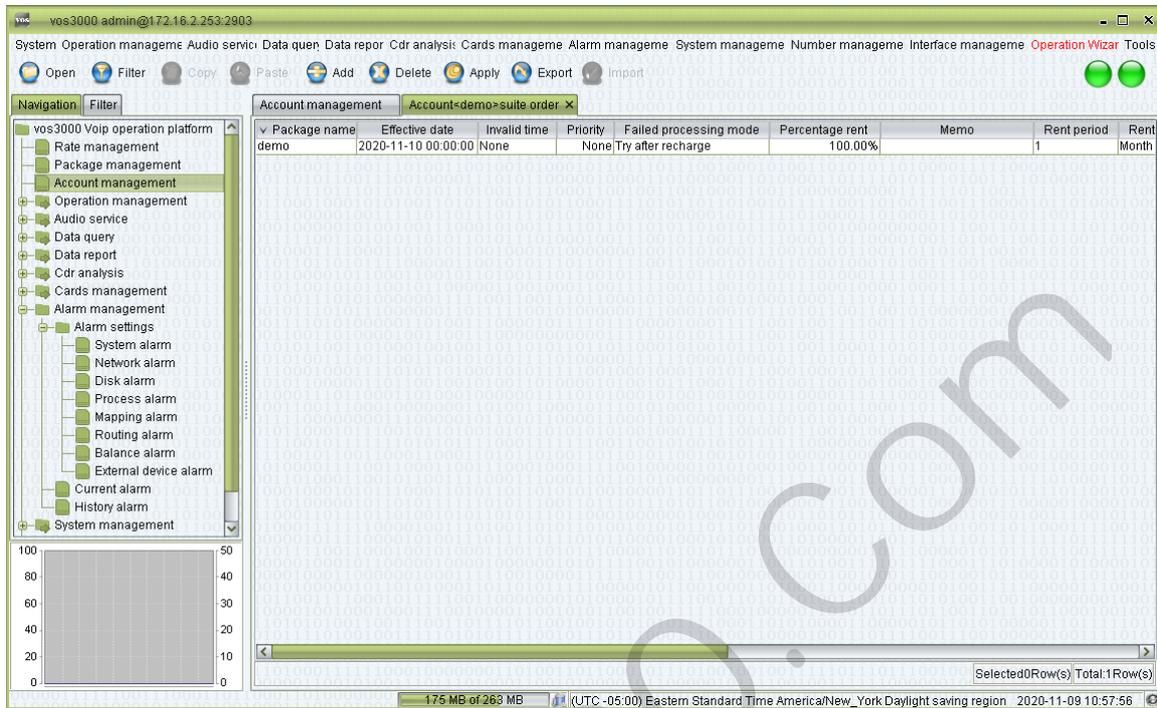
NOTE

If an account is deleted, his phone and gateway will belong to no account and will not be deleted by association.

You cannot change clearing account into other type.

2.4.1.1 Customer Package Management

This function is used to manage accounts packages.



How to Start

- Double-click “Navigation > Account management > Suite order”

Table Items

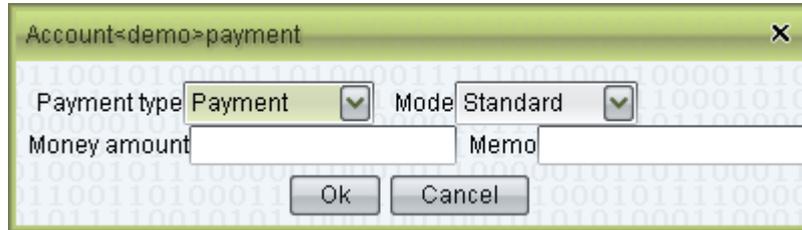
- Package name: name of the package need to order.
- Effective date: time of the package begin order.
- Invalid time: time of the package stop order.
- **NOTE**
- Add packages for the account then choose package name and set the effective date and invalid time of packages.
- The “Invalid time” is the date after which the package cannot be subscribed. For example, if a monthly lease has effective date (2013-9-2) and invalid time (2013-9-15), on the 2th of September; the package will be subscribed, and effective through 2013-9-2 to 2013-9-30. But after September 15th, the package will not be available for subscription. That means this package cannot be used in October.
- Priority : package with high priority will take effect firstly (the smaller the value, the higher the priority), if packages' priority are the same but not “none”, account balance will determine packages together to take effect or not. priority with "none" will finally take effect. if packages' priority are all "none", system will determine the effective order of package.
- Failed processing mode: the processing mode after suite order failed.
 - Try after recharge: immediately try to order suite again once account be recharged.
 - Try next cycle: try to order suite again in next cycle.
 - Delete order: delete the order directly.
- Percentage rent: rent of percentage.

- Memo: comments on the package.
- Rent period, Rent unit, Rent fee, Minimum consumption, Become effective spending limit, Period rate, Free duration, Free money amount, Suite memo: display package information corresponding to the package name after the apply succeeds.

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2.4.2 Payment

This function is used to pay money for account.



How to Start

- Double-click “Navigation > Account management”, select an account; click “Payment” in the right-click menu.

Operation Details

- Type of payment: If the phone card is selected, the card number and password will be asked.
- Payment type: Payment/Credit/Make Zero.
- Amount: the amount of payment.
- Memo: comments will be kept as historical records.



NOTE

Historical payments can be view in the payment record page.

Payment will change account's expiration. Please refer to system parameter:
SERVER_PAY_DELAY_CUSTOMER_EXPIRE_DAY

2.4.3 Agent Account

Agent accounts differ with ordinary accounts in that there are accounts belonging to agent accounts. Once an account becomes an agent account, it will occur in the navigation tree. Double-click the agent account in the navigation tree to open the “Sub account management”.



NOTE

Use the filter “Direct” and “All”, respectively, to show the directly sub accounts and all (direct and indirect) affiliations of the account.

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2.4.4 Billing

Phones, gateways and bind numbers will be charged according the account they currently belong to. If the number being called does not match any rate, the call will be terminated. And the cause of such termination will be shown in CDR. When the account belongs to other accounts, the call will cause the agent account to be charged according to its own rate (this backtracking process ends up at accounts that belong to no other accounts). If the billing turns the account or any of its agent accounts into “disabled” status, the phones, gateways and bind numbers will no longer be able to make calls.



NOTE

Billing principle: optimal rate, the deduction amount is calculated by period fee rate, account fee rate, account private fee rate or phone private fee rate, choose the cheapest.

When account is in debt, he can still make free call.

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2.4.5 Authorization Management

This function is used for manage agent's operation of accounts.



How to Start

- Double-click “Navigation > Account management”, select an account, click “Authorizations” in the right-click menu

Operation Details

- Authorizations
- Add/delete/modify account: the right to create delete or modify accounts.
- Add/delete/modify phone: the right to manipulate phones belonging to the account.
- Add/delete/modify phone card: the right to manipulate phone cards belonging to the account.
- Add/delete gateway: the right to manipulate gateways belonging to the account.
- Modify gateway information: the right to modify information about gateways except capacity.
- Modify gateway capacity: the right to modify the number of lines.
- Payment for this account: the right to perform payment for the current account (including changing the amount of overdraft).
- Payment for sub accounts: the right to pay for the sub accounts.



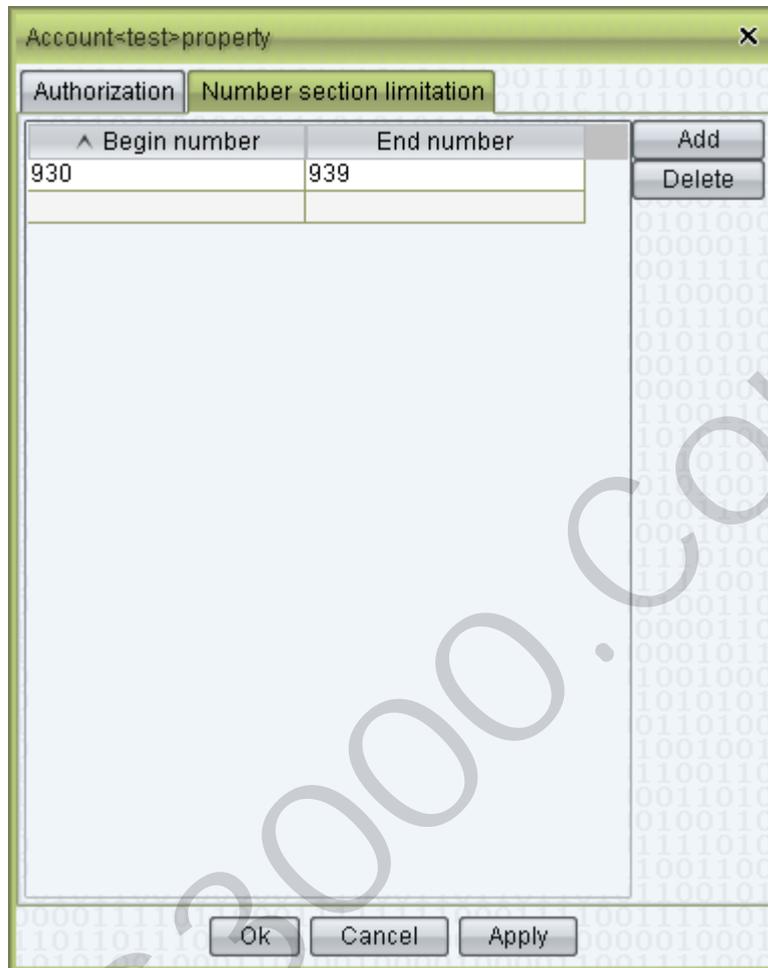
NOTE

This function is usually used to facilitate agent development. An agent user can have an agent-typed account in the system. Administrators can create one or more accounts for them, limiting their rights to recharge their own account, yet granting those authorizations to add new accounts, phones and gateways, and to recharge their sub accounts. The agent can create new accounts for its sub-agents. The agent account can only manipulate its sub accounts. Note that, accounts created by agent accounts must be designated to an agent account, and the creator must have the authorization to manipulate the designated agent account. Users logged in with an agent account can only see those accounts that authorized to the agent. This restriction applies to all account-related operations.

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2.4.6 Number Section Limitation

This function is used to limit the phone prefix of the account.



How to Start

- Double-click “Navigation > Account management”, select an account, click “Authorizations” in the right-click menu.

Operation Details

- Begin number: the smallest number of the segment (including the number itself).
- End number: the largest number of the segment (including the number itself).



NOTE

Number restriction specifies the phone numbers that can be added to the current account, in order to avoid competition of number resources among agents.

If account type is “agent”, the numbers of its sub accounts should also be in this range. Otherwise there will be error prompt from the system. If the account type is “ordinary”, the appropriate number segments will automatically added by the system.

2.4.7 Modify Cdr

This function is used to modify the charging amount of the account history bill.



- Begin time: begin time of bill revision.
- End time: end time of bill revision.
- Time zone: fix the time zone of the bill time.
- Billing rate: the rate of correcting the bill charge amount.

 **NOTE**

When the charging rate is the default, the modified bill rate is modified according to the billing rate of the account.

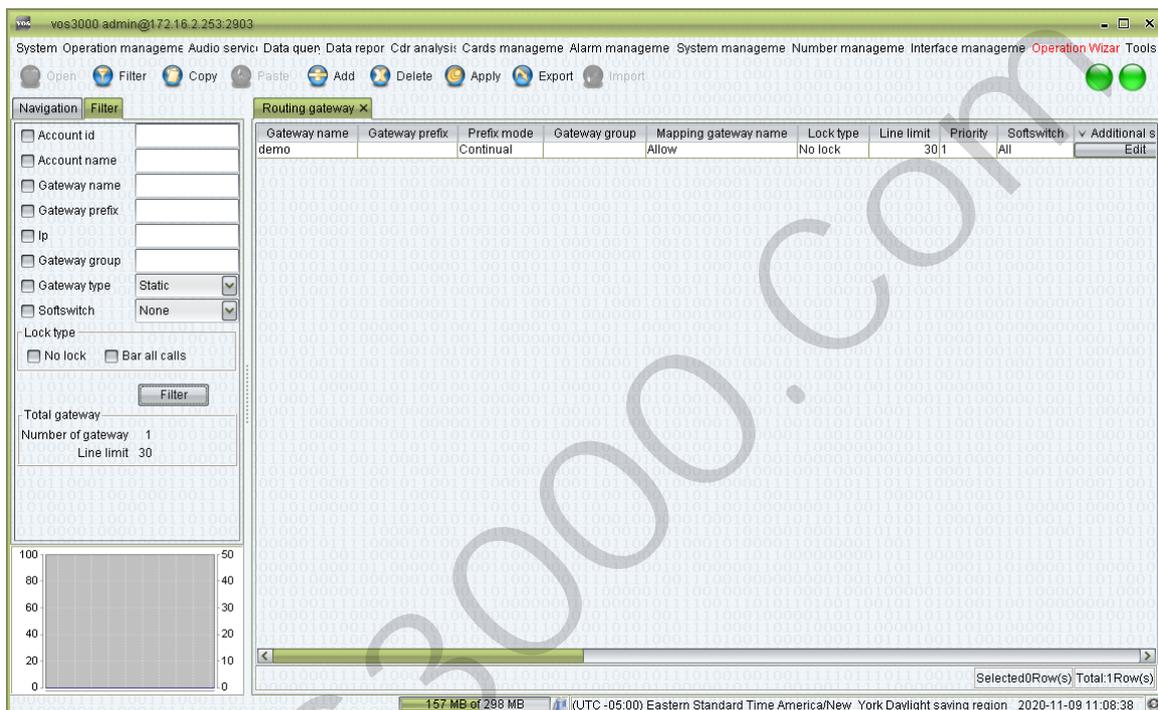
The bill rate of the settlement account cannot be modified to correct the bill rate of the settlement account.

2.5 Operation Management

2.5.1 Gateway Operation

2.5.1.1 Routing Gateway

This function is used to manage routing gateway.



How to Start

- Double-click “Navigation > Operation management > Gateway operation > Routing gateway”

Table Items

- Gateway name: the unique id of the device, used for the authentication of dynamic gateways. For static gateways (usually relay gateways), the only requirement is their ids do not conflict with each another.
- Gateway prefix: when the number being called is not registered in the system, the call will be routed only to gateways which match the prefix specified here. Multiple prefixes can be specified, separated by commas. Different gateways can be designated with the same prefix. When conflict occurs, the gateway will be chosen according to following numbers (the smallest comes first): priority number, the ratio of the number of current calls to the number of channels, the number of historical calls, and the gateway id.
- Prefix mode:
 - Extension: shorter prefixes will be tried if the routing gateway matched by this prefix cannot deliver the call.

- Expiration: no more prefixes will be tried if the routing gateway matched by this prefix cannot deliver the call.

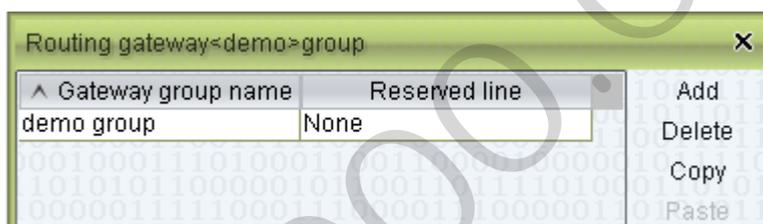
Examples:

Gateway id	Gateway prefix	Prefix mode	Gateway group	Lock type	Line limit	Priority
gw1	9	Continual		No lock	30	1
gw2	900	Terminal		No lock	30	0
gw3	9	Continual		No lock	30	1
gw4	900	Continual		No lock	30	1

If the prefix mode of “gw2” is set to “Terminal”, the prefixes being tried for the number “90080001” will be “gw2” and “gw4” in order.

If the prefix mode of “gw2” is set to “Continual”, while others remain the same, the prefixes being tried for the number “90080001” will be “gw2”, “gw4”, “gw3”, and “gw1” in order.

- Gateway group: the group which gateway belongs to.
- Reserved line: line at least to be reserved for gateway group. Group line limit will restrict gateway line limit when this gateway belong to this group. If remaining available line of group is below the value of reserved line, this gateway will be disabled. Means when line resource strained, function of reserved line used to assure line resource is available for important customers.



NOTE

Suppose that client A and client B use the same routing gateway, which has a line limit of 600(client A and client B have different callee prefixes). If we want to give client A and client B different priorities as : client A is permitted to use all the 600 lines; client B is permitted to use only 200 lines for the most; if client A's concurrency is over 550 lines, the routing gateway shuts down the service for client B and only provide service for client A, we can do as the following.

Create gateway group G, Line limit is 600.

Create routing gateway A which only allow client A to pass through, line limit is 600, belonging to group G and the reserved line is “none”.

Create routing gateway B which only allow client B to pass through, line limit is 200, belonging to group G and the reserved line is “50”.

According to the above configuration, when call can use the gateway A, concurrent gateway group G can reach 600 (Gateway Group line limit), when call can use the gateway B, concurrent gateway group G can not exceed 550 (Gateway Group line limit minus 50 lines), If the gateway A current concurrency is 520, the gateway B current concurrency is 40, the call can use the gateway A, and can not use the gateway B. From another point of view can be considered, at the peak of the priority to protect the ability to connect the client A, and in control of the maximum-peak concurrent client B's.

Note that this ground if the overall concurrent gateway less than 550, the maximum concurrent client B can reach 200. Client A and client B who call first, only to restrict client B calls, when gateway concurrent more than 550.

NOTE

Instructions for combination use of period capacity and reserved line

Create gateway group G, line limit is “none”.

Create routing gateway A, line limit is “none”, belonging to G and reserved line is “none”, gateway A's period capacity is that during00:00:00-18:00:00, line limit is 200, during 18:00:00-24:00:00, line limit is 400.

Create routing gateway B, line limit is 200, belonging to G and reserved line is 50.

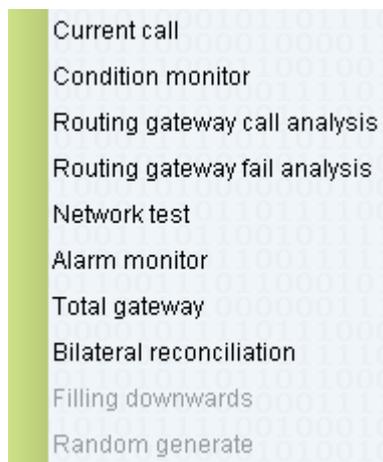
As above setting, during 00:00:00-18:00:00 every day, when gateway A is available, G's concurrency can reach 400 when gateway B is available, G's concurrency can reach 350, during 18:00:00-24:00:00 every day, when A is available, G's concurrency can reach 600. When B is available, G's concurrency can reach 550.

- Mapping gateway name: Set the mapping gateways which are allow/forbidden to use the routing gateway.
- Lock type: "No lock" or "Bar all calls".
- Line limit: lines of this routing gateway.
- Priority: when the prefixes are the same, gateway with high priority will be used firstly.
- Softswitch: specify the softswitch used by this routing gateway.
- Additional settings
- Ip: see descriptions in the "Additional settings".
- Configuration password: the password used for gateway registration, which is also the password used for web configuration.
- Self service password: the password used for web query.
- Caller black/white list group: can set numbers in "Number management > Black/white list group".
- Memo: comments on the gateway.
- Routing clearing account id: the billing account charged when the gateway is called.
- Routing clearing account name: the name of the billing account.
- Routing clearing account balance: the balance of the billing account.
- Clearing billing number:
 - Before change dial plan: consistent with this call's billing number.
 - After change dial plan: using the number of the dial plan rewritten by this routing gateway configuration.

Other Operations

- Double-click the "Routing clearing account name", can directly enter to the manage interface of this account.

Right-Click Menu

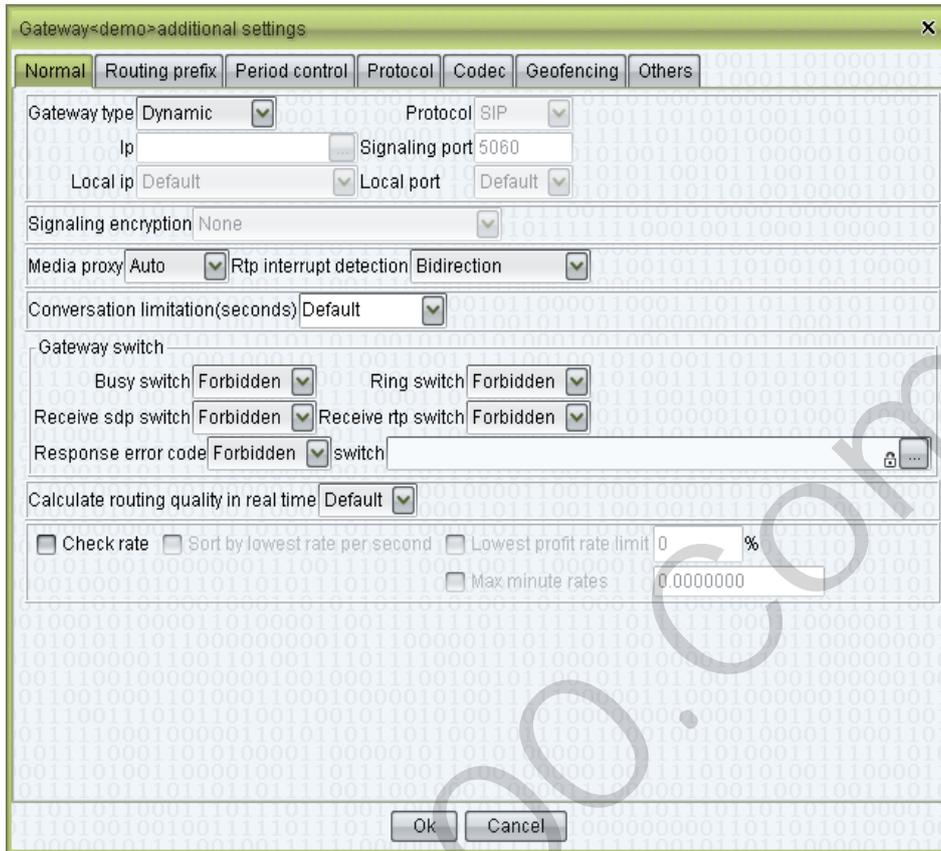


- Current call: open the "Current call" list of the gateway.

- Condition monitor: monitor status of the gateway and view details in “Gateway status”.
- Routing gateway call analysis: open “Connect analysis” page of the gateway.
- Routing gateway fail analysis: open “Interrupt analysis” page of the gateway.
- Network test: test the network of to-end.
- Alarm monitor: open “Alarm monitor” page of the gateway.
- Total gateway: count the total number of multiple gateways.
- Bilateral reconciliation: reconciliation between two platform accounts.
- Filling downwards: use for copy information.
- Random generate: generate random password of newly added gateways.

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Additional settings > Normal



- Gateway type:
 - Static: no registration is required. These are usually relay gateways (i.e. routing gateway). Static IP addresses and ports can be specified for them.
 - Dynamic: registration is required before use.
 - Registration: register to other server; please refer to “Registration management”.
- Protocol: H323 or SIP.
- Ip: gateway's IP.
- Signaling port : gateway's port.
- Local ip: set IP of local network used for sending, use “Auto” to choose by Linux routing table.
- Local port: set port of local network used for sending
- Media proxy:
 - Auto: let the system decide whether enable media proxy. (Recommended)
 - On: always enable media proxy.
 - Off: always disable media proxy.
 - Must on: media forwarding must be turned on.
- RTP interrupt detection:
 - None: disable detection.
 - Server to remote: detect audio send from server to device.
 - Remote to server: detect audio send from device to server.

- Bidirection: detect both side, if any one side no audio, the call will be interrupt.
- Conversation limitation (seconds): set the max call duration of the gateway.
 - None: no limit.
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_MAX_CALL_DURATION”.
- Switch gateway until connect:
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_GATEWAY_SWITCH_UNTIL_CONNECT”.
 - On: switch gateway until softswitch got connect signal (SIP 200 OK/ H323 Connect).



NOTE

This option priors to “Protocol > Stop switch gateway after olc” and “Stop switch gateway after receive sdp”.

- Off: try next gateway except: call connected, ringing, receive busy or no answer and settings in “Protocol”.
- Stop switch gateway when rtp start:
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_GATEWAY_SWITCH_STOP_AFTER_RTP_START”.
 - On: if using media proxy, when softswitch received RTP packet from the gateway, won't try next gateway any more.



NOTE

This option is NOT affected by “Switch gateway until connect”. When “Switch gateway until connect” is on, if received RTP packet, stop switch gateway.

- Off: ignore the RTP starting conditions for gateway switch.
- Callee busy stop switch:
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_GATEWAY_SWITCH_STOP_AFTER_USER_BUSY”.
 - On: when softswitch received busy signal from the gateway, won't try next gateway any more.



NOTE

This option is NOT affected by “Switch gateway until connect”. When “Switch gateway until connect” is on, if received busy signal, stop switch gateway.

- Off: ignore the RTP starting conditions for gateway switch.
- Calculate routing quality in real time:
 - Default: determined by “Softswitch Management > Additional set > System parameter > SS_GATEWAY_QUALITY_CALCULATE” parameter.
 - On: softswitch calculates the routing quality of this routing gateway in real time.
 - Off: softswitch does not calculate the routing quality of this routing gateway in real time.
- Check rate: if the call has clearing fee rate, this gateway will be tried.
- Sort by lowest rate per second: use rate per second when sorting. Sort order can be set in “Operation management > Softswitch management > Additional settings > System parameter > SERVER_GATEWAY_ROUTE_FEE_RATE_BEFORE_QUALITY” and “SERVER_GATEWAY_ROUTE_FEE_RATE_SORT_CONFIG”.
 - Enable: during fee rate sorting, use actual fee rate.

- Disable: during fee rate sorting, see the gateway has the lowest fee rate.
- Lowest profit rate limit: lock this gateway when profit below settings. When the difference, calculate by rate per second, between caller fee rate and clearing fee rate lower than the value, this gateway won't be tried.



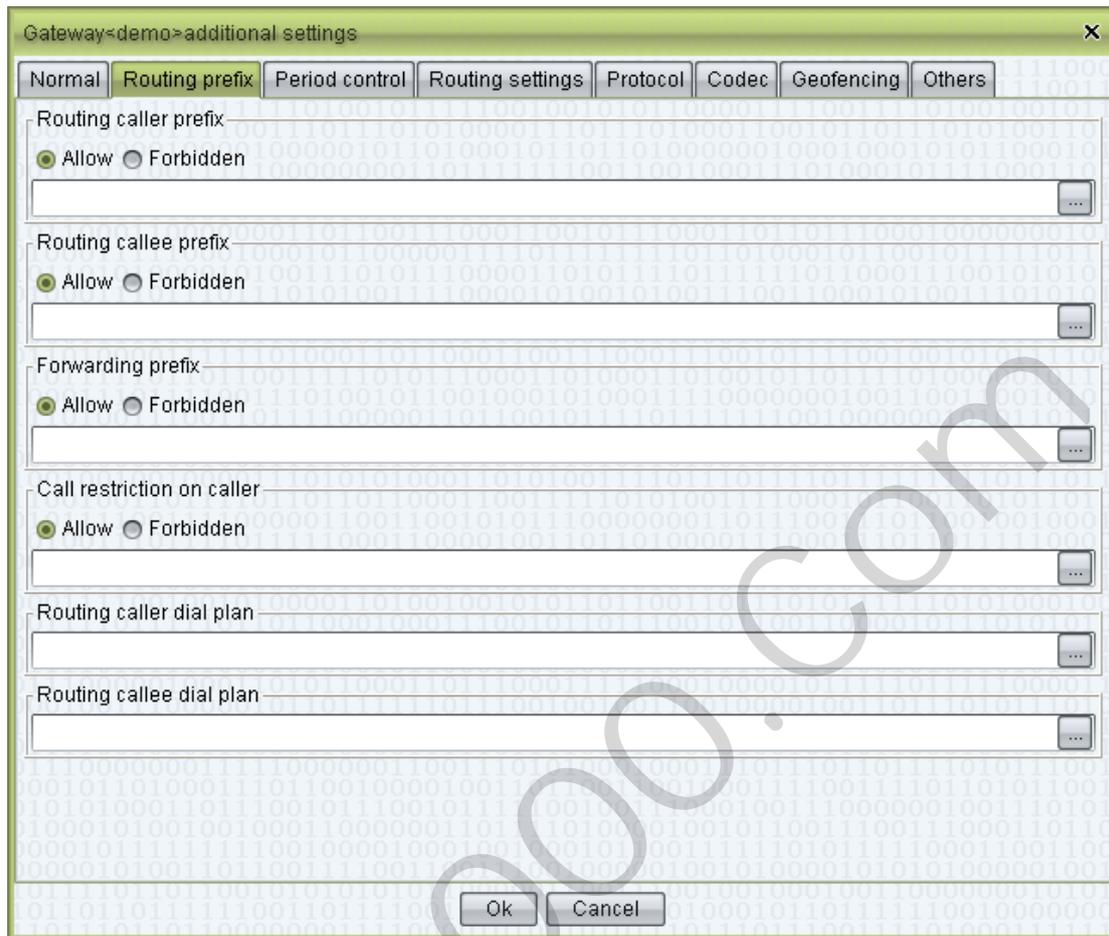
NOTE

Negative is supported.

- Max minute rates: when minute rate above the value, this gateway won't be tried.

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Additional settings > Routing prefix



- Routing caller prefix:
 - Allow: prefixes of the caller numbers allowed to pass through (left blank to allow all numbers).
 - Forbidden: prefixes of the caller numbers disallowed to pass through.

NOTE
Only one of the “Allow” and “Forbidden” options can be chosen.

- Routing callee prefix:
 - Allow: prefixes of the called numbers allowed to pass through.
 - Forbidden: prefixes of the called numbers disallowed to pass through.
- Forwarding prefix:
 - Allow: forwarding prefixes to pass through (left blank to allow all numbers).
 - Forbidden: disallow forwarding prefixes to pass through.

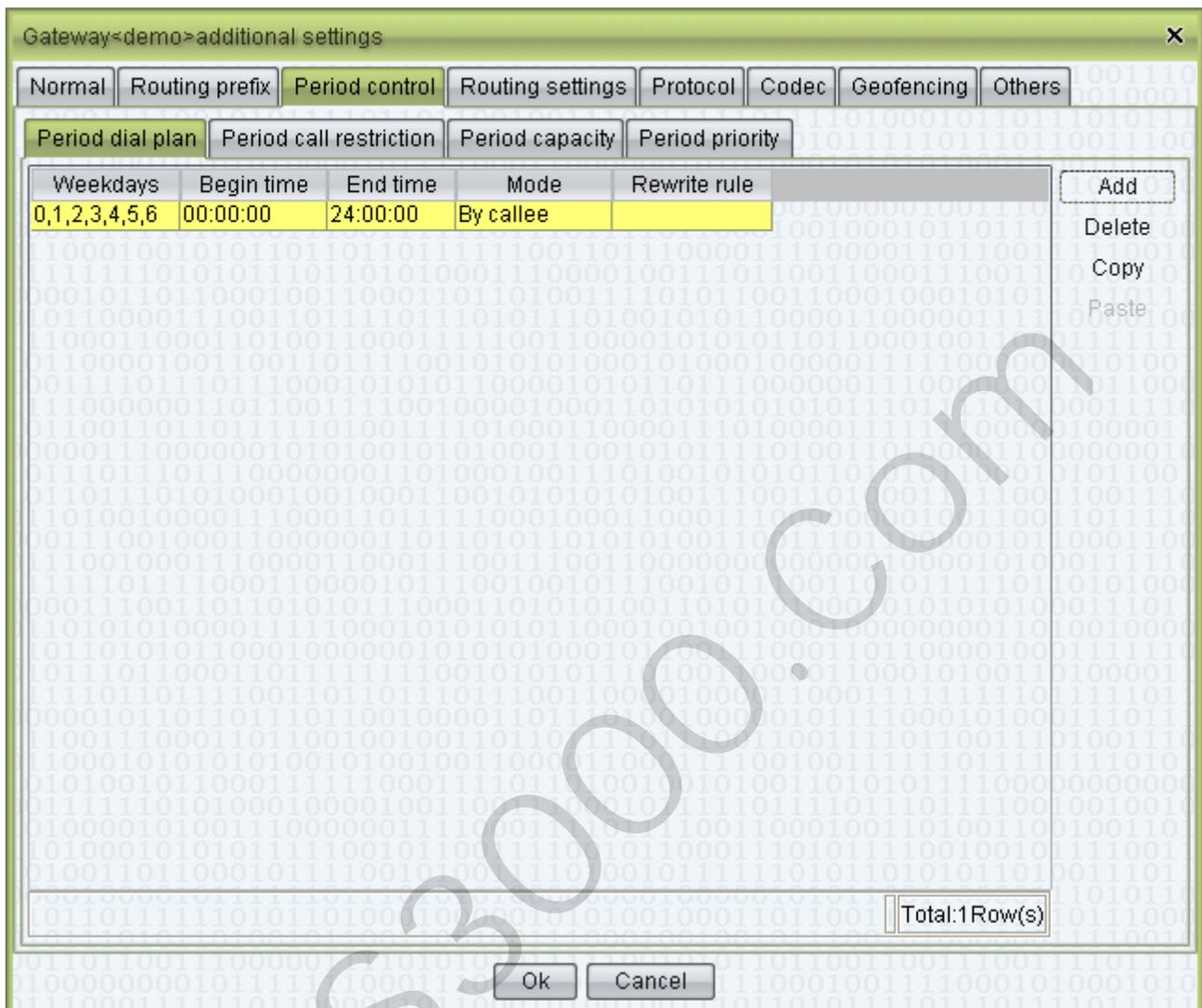
NOTE
Forwarding number is in the To field of forwarding signal.

- Call Restriction on Caller:
 - Allow: allow the caller with particular prefix to dial callee with particular prefix.
 - Forbidden: forbidden the caller with particular prefix to dial callee with particular prefix.

- Routing caller dial plan: change dial plans for the caller number when called out through this gateway.
- Routing callee dial plan: change dial plans for the called number when called out through this gateway.

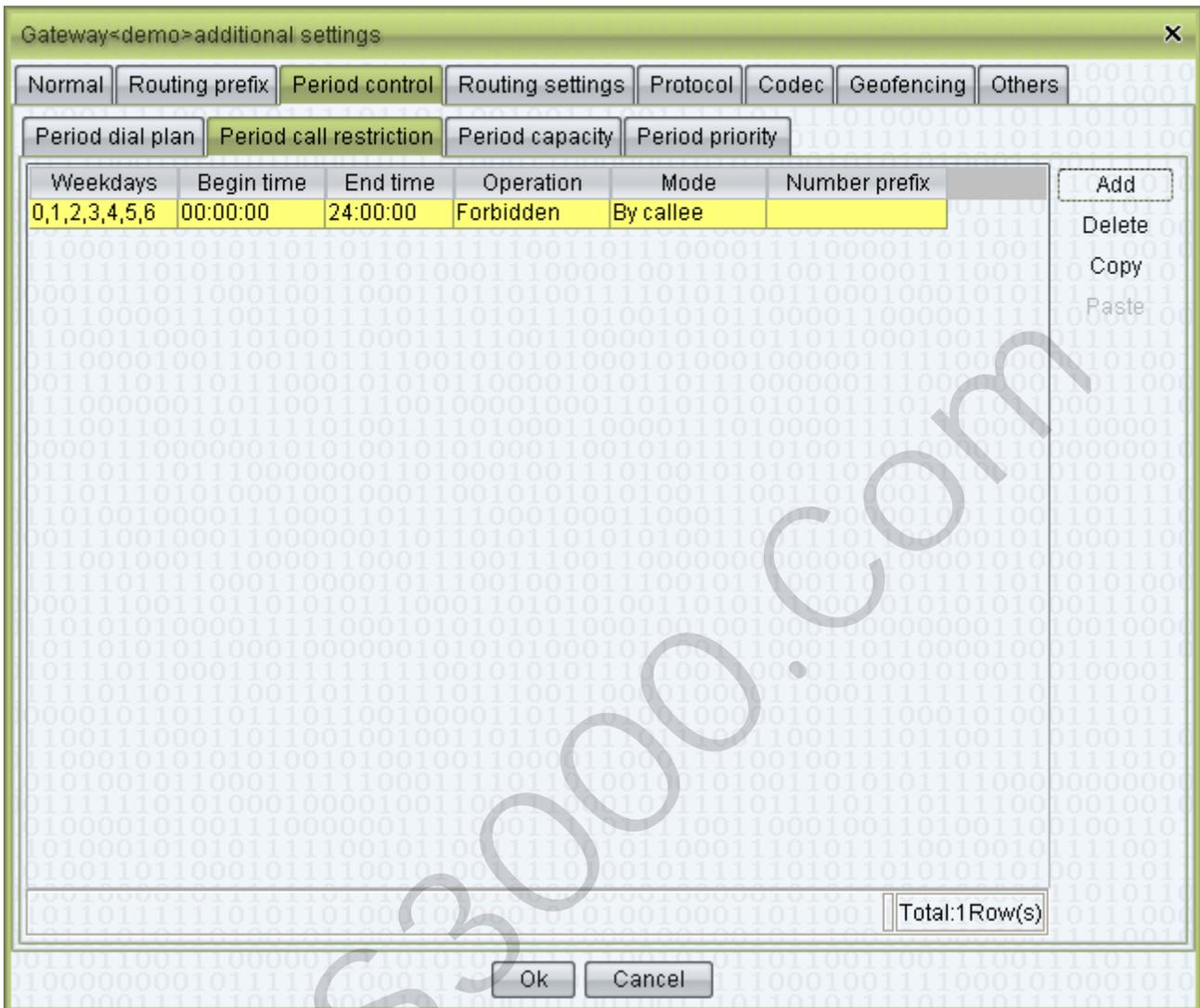
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Additional settings > Period control > Period dial plan



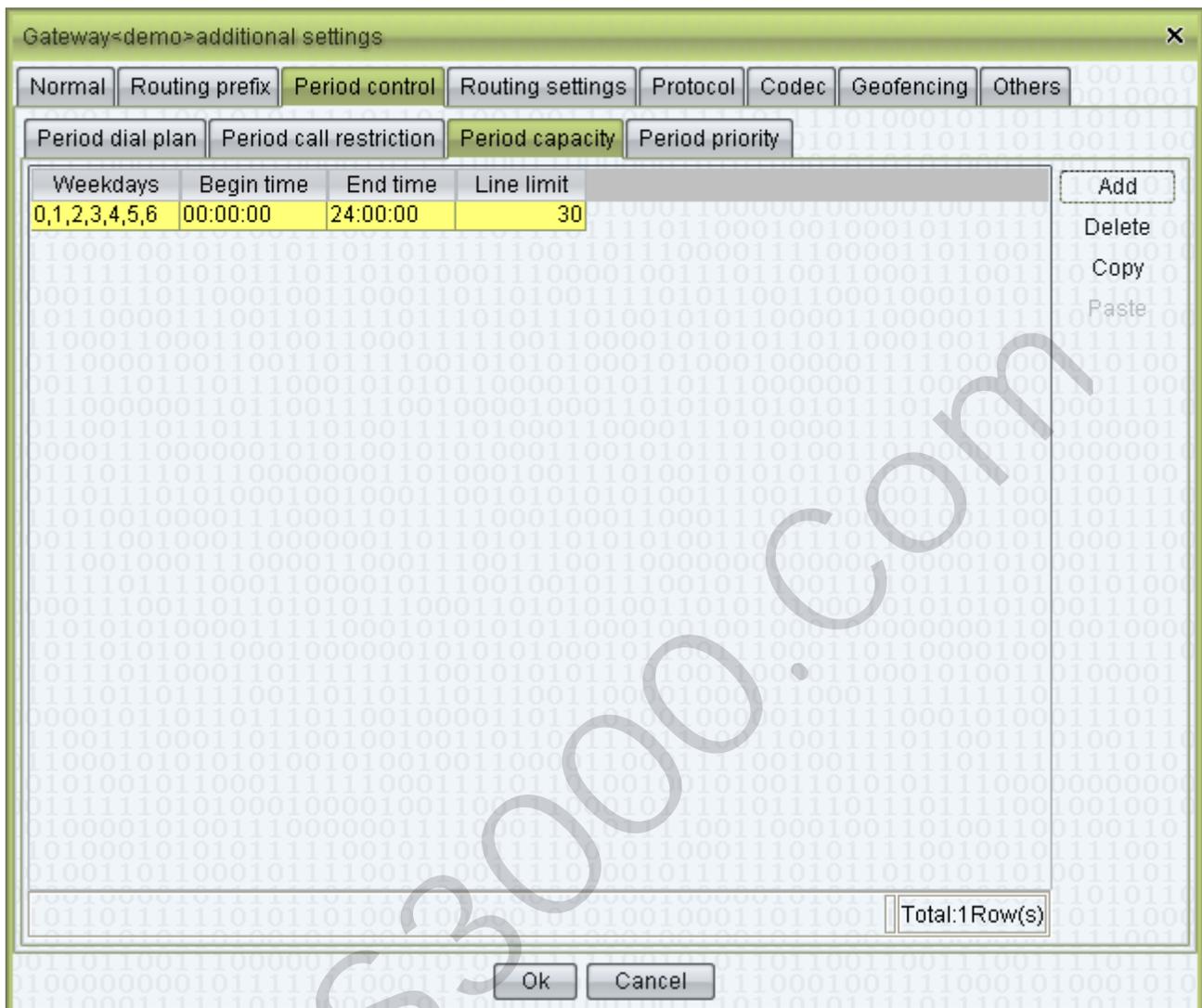
- Weekdays: time corresponding to the week.
- Begin time: time when the dial plan comes into effect.
- End time: time when the dial plan expires.
- Mode:
 - The called: the dial plan applies to the called number.
 - The caller: the dial plan applies to the caller number.
- Dial plan: the content of the rule.

Additional settings > Period control > Period call restriction



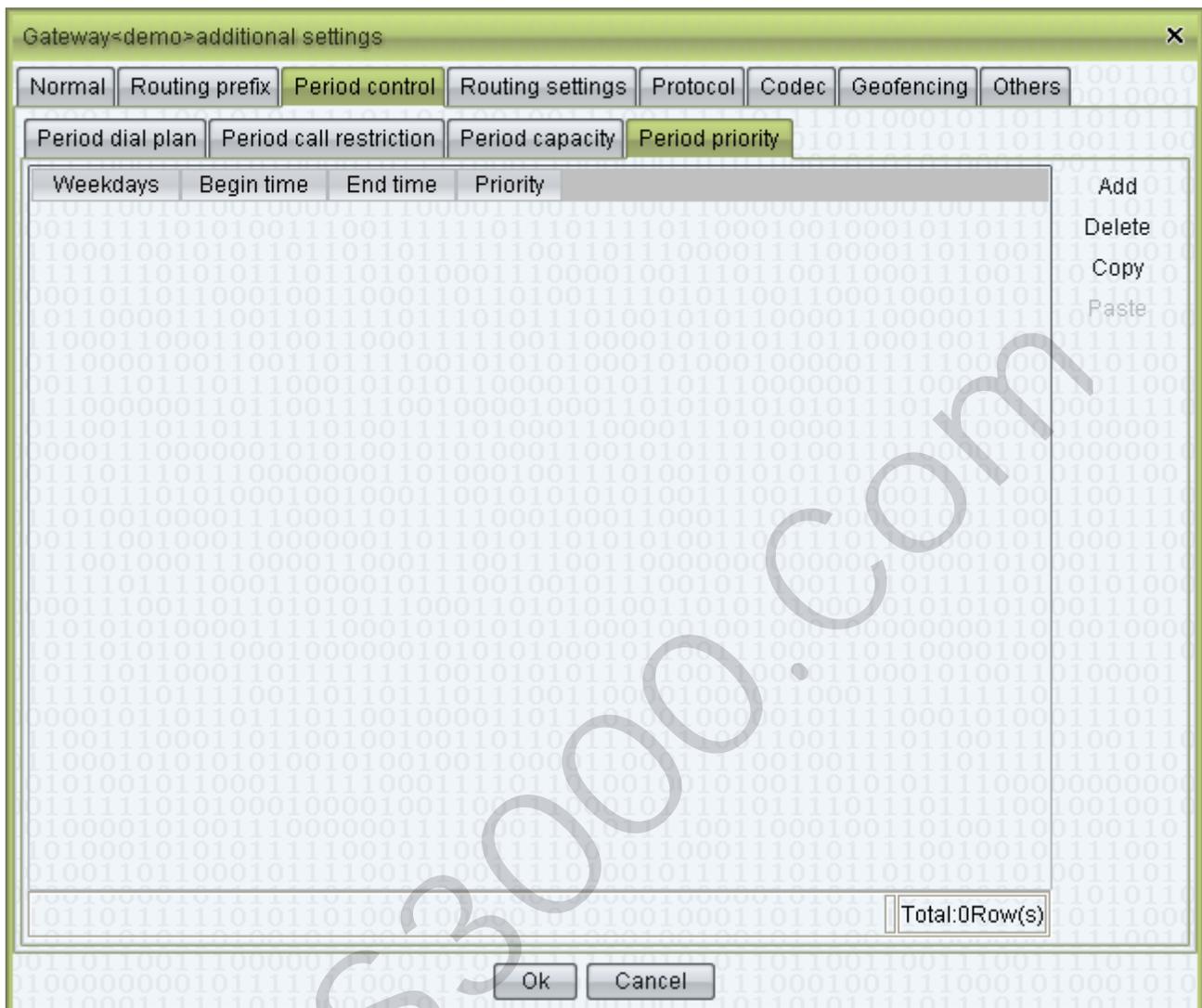
- Weekdays: time corresponding to the week.
- Begin time: time when the rule comes into effect.
- End time: time when the rule expires.
- Operation:
 - Forbidden: forbidden operations for the matched prefixes.
 - Allow: allow operations for the matched prefixes.
- Mode:
 - By callee: matches the prefixes of the callee numbers.
 - By caller: matches the prefixes of the caller numbers.
- Number prefix: the prefix of the number. Multiple prefixes can be specified, separated by commas.

Additional settings > Period control > Period capacity



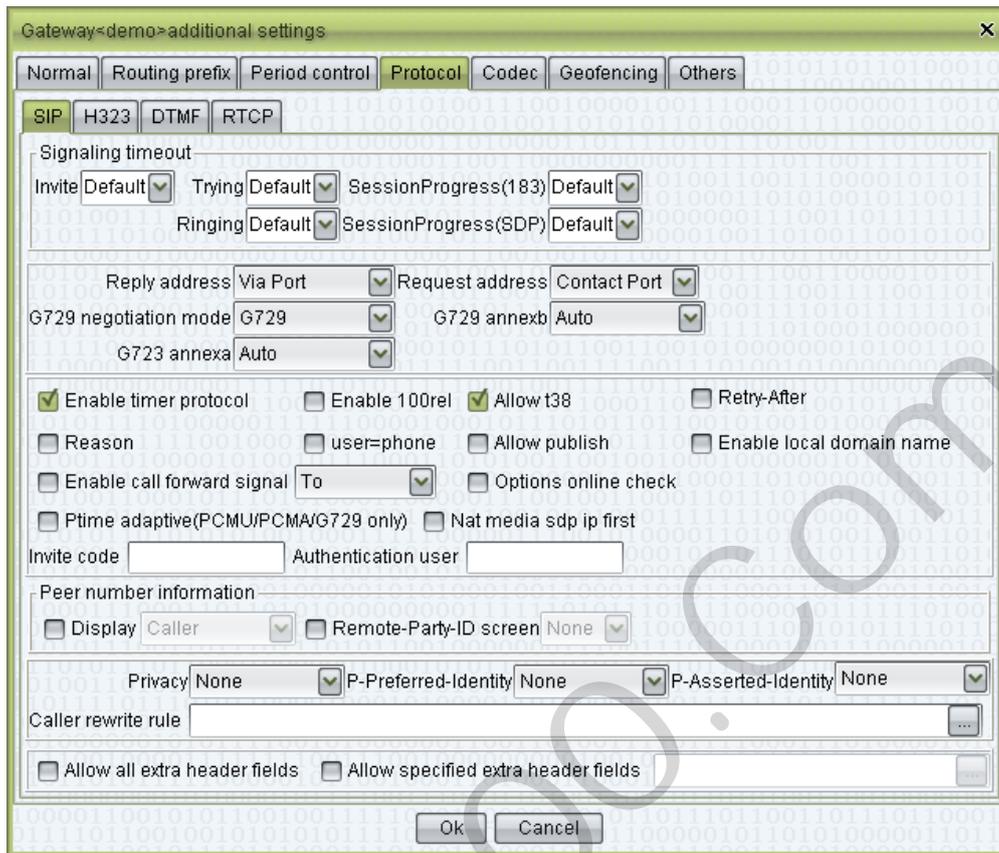
- Weekdays: time corresponding to the week.
- Begin time: time when the rule comes into effect.
- End time: time when the rule expires.
- Line limit: the maximum number of channels allowed for this period.

Additional settings > Period control > Period priority



- Weekdays: time corresponding to the week.
- Begin time: time when the rule comes into effect.
- End time: time when the rule expires.
- Priority: priority for this gateway for this period.

Additional settings > Protocol > SIP



NOTE

Default value is set in “Operation management > Softswitch management > Additional settings > SIP parameter”.

- Invite/Trying/Ringing/SessionProgress(183)/SessionProgress(SDP): if signal timeout, softswitch will try next gateway or hang up.
 - Invite default: set by” Softswitch management > Additional settings > SIP parameter > SS_SIP_TIMEOUT_INVITE”.
 - Trying default: set by” Softswitch management > Additional settings > SIP parameter > SS_SIP_TIMEOUT_TRYING”.
 - Ringing default: set by” Softswitch management > Additional settings > SIP parameter > SS_SIP_TIMEOUT_RINGING”.
 - SessionProgress(183) default: set by” Softswitch management > Additional settings > SIP parameter > SS_SIP_TIMEOUT_SESSION_PROGRESS”.
 - SessionProgress(SDP) default: set by” Softswitch management > Additional settings > SIP parameter > SS_SIP_TIMEOUT_SESSION_PROGRESS_SDP”.
- Stop switch gateway after receive sdp:
 - Off: won’t affect gateway switch.
 - On: stop switch gateway after receive SDP.
 - Default: set by” Softswitch management > Additional settings > SIP parameter > SS_SIP_STOP_SWITCH_AFTER_SDP”

NOTE

If “Switch gateway until connect” is on, this setting is invalid.

- Stop switching response code: stop switch gateway when receive this code.
- Reply address: after receive sip request message, send reply signal to which address.
 - Socket: send reply signal to request address (recommend).
 - Via port: send reply signal to request IP and use port from Via. (Due to network reasons, some systems will use this mode).
 - Via: send reply signal to address from Via. (May have security risk).
- Request address: after call establish, send request signal to which address.
 - Socket: send request signal to sender address (recommend).
 - Contact Port: send request signal to sender IP and use port from Contact.
 - Contact: send request signal to address from Contact.
- G729 negotiation mode:
 - Auto: keep original G729 codec.
 - G729: treat G729a or G729 as G729.
 - G729a: treat G729 or G729a as G729a.
 - G729&G729a: treat G729 or G729a as G729 and G729a.
- G729 annexb:
 - Auto: send routing's G729 annexb setting to routing.
 - yes: annex=yes.
 - no: annex=no.
 - None: no annexb.
 - Passthrough: send caller's G729 annexb setting to routing.
- G723 annexa: refer to G729 annexb.
- Enable timer protocol: enable routing gateway's timer protocol to detect over time.
- Enable 100rel: send 1xx to routing gateway (intermediate state like 183).
- Allow t38: allow send T38 signal to routing gateway.
- Retry-After: when routing gateway's signalling contain Retry-After domain, the values will determine the disabled length of gateway, disabled gateways are displayed in "Online routing gateway".
- Reason: add reason head to the call's hanging signaling (cancel, bye, or direct rejection of the caller's error response) for the transmission to the end of the reason head (when the end of the hanging signaling contains the reason head), or the call hanging description that contains VOS.
- user=phone: add the User=phone field to invite of call request signaling.

```

INVITE sip:13812345678@172.16.5.36;user=phone SIP/2.0
via:SIP/2.0/UDP 172.16.4.182:5060;branch=z9hG4bK61c15f013ea537e4
From:"70013" <sip:70013@172.16.4.182;user=phone>;tag=732f16783a01121f
To:<sip:13812345678@172.16.5.36;user=phone>
P-Charging-vector:cid-value="TJmNDVAMmRhNjM2MTZINmMowMDAwMDE1MUANZluMTYuN
C4xODKuKwde";orig-ioi="";term-ioi=""
Remote-Party-ID: "70013" <sip:70013@172.16.4.182>
P-Preferred-Identity: <sip:70013@172.16.4.182;user=phone>
P-Asserted-Identity: <sip:70013@172.16.4.182;user=phone>

```

Call-ID: Y2145d2da63616c6coo000151@172.16.4.182

cseq: 21521 INVITE

Contact: <sip:70013@172.16.4.182:5060>

Allow: INVITE,ACK,CANCEL,BYE,OPTIONS,INFO,UPDATE,PRACK

- Allow Publish: this protocol can make routing gateway control concurrency automatically.
- Enable local domain name: change dial plan the IP corresponding to the “from” field in signaling to SS_LOCAL_IP_DOMAIN in “softswitch system parameters” corresponding to domain.
- Enable call forward signal: individual customization function, please contact technical support for further information.
- Options online check: according to the soft switch system parameter “SS_SIP_OPTIONS_CHECK_PERIOD” means sending SIP OPTIONS messages periodically to detect whether the SIP port of the routing gateway is reachable. When OPTIONS detection fails, it will switch to the IP port with successful OPTIONS detection. If all IP and ports fail, the routing gateway is impossible until the subsequent OPTIONS detection is successful.
- Ptime adaptive: when media proxy is on, ensure that the RTP packaging cycle sent to the peer is consistent with the Ptime packaging cycle of the receiving peer.
- Nat media sdp ip first: when returning to RTP, the SDP address of media is preferred.

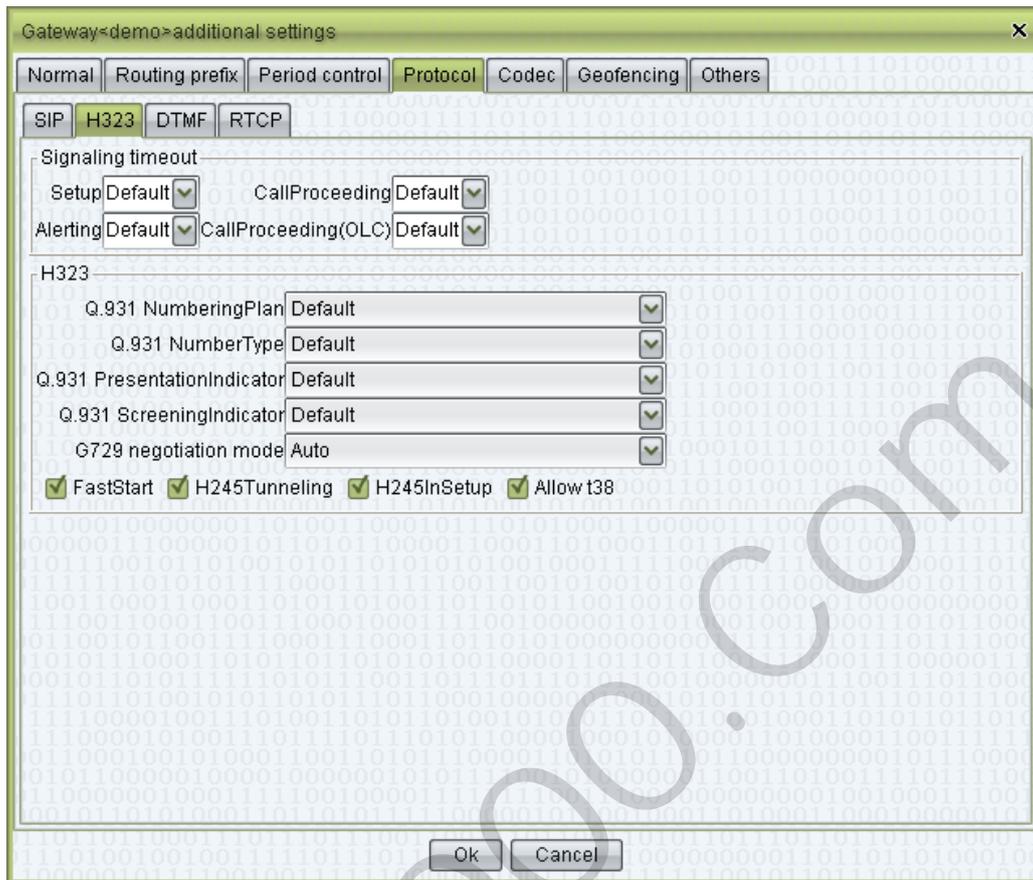


NOTE

After enable call forward signal, the outbound caller of CDR will be “number in To field | number in From field”.

- Invite code: add a custom field to the invite field.
- Authentication user: users authenticated.
- Peer number information: set select mode to SIP signal’s caller.
- Privacy:
 - None: no this field.
 - Passthrough: send received settings to routing gateway.
 - Id: privacy field.
- P-Preferred-Identity:
 - None: no this field.
 - Passthrough: send received settings to routing gateway.
 - Caller: use caller number.
- P-Asserted-Identity:
 - None: no this field.
 - Pass through: send received settings to routing gateway.
 - Caller: use caller number.
- Caller dial plan: dial plans for the caller number in “P-Asserted-Identity” field.
- Allow all extra header fields: SIP header transparent, allowing all additional header domains.
- Allow specified extra header fields: Additional header fields can be added and specified independently.

Additional settings > Protocol > H323



NOTE

Default value is set in "Operation management > Softswitch management > Additional settings > H323 parameter".

- Setup/CallProceeding/Alerting/CallProceeding(OLC): if signal timeout, softswitch will try next gateway or hang up.
 - Setup default: set by "Softswitch management > Additional settings > H323 parameter > SS_H323_TIMEOUT_SETUP".
 - CallProceeding default: set by "Softswitch management > Additional settings > H323 parameter > SS_H323_TIMEOUT_CALLPROCEEDING".
 - Alerting default: set by "Softswitch management > Additional settings > H323 parameter > SS_H323_TIMEOUT_ALERTING".
 - CallProceeding(OLC) default: set by "Softswitch management > Additional settings > H323 parameter > SS_H323_TIMEOUT_CALLPROCEEDING_OLC".
- Stop switch gateway after olc:
 - Off: won't affect gateway switch.
 - On: stop switch gateway after receive faststart or OpenLogicalChannel (H245).
 - Default: set by "Softswitch management > Additional settings > H323 parameter > SS_H323_STOP_SWITCH_AFTER_OLC".

NOTE

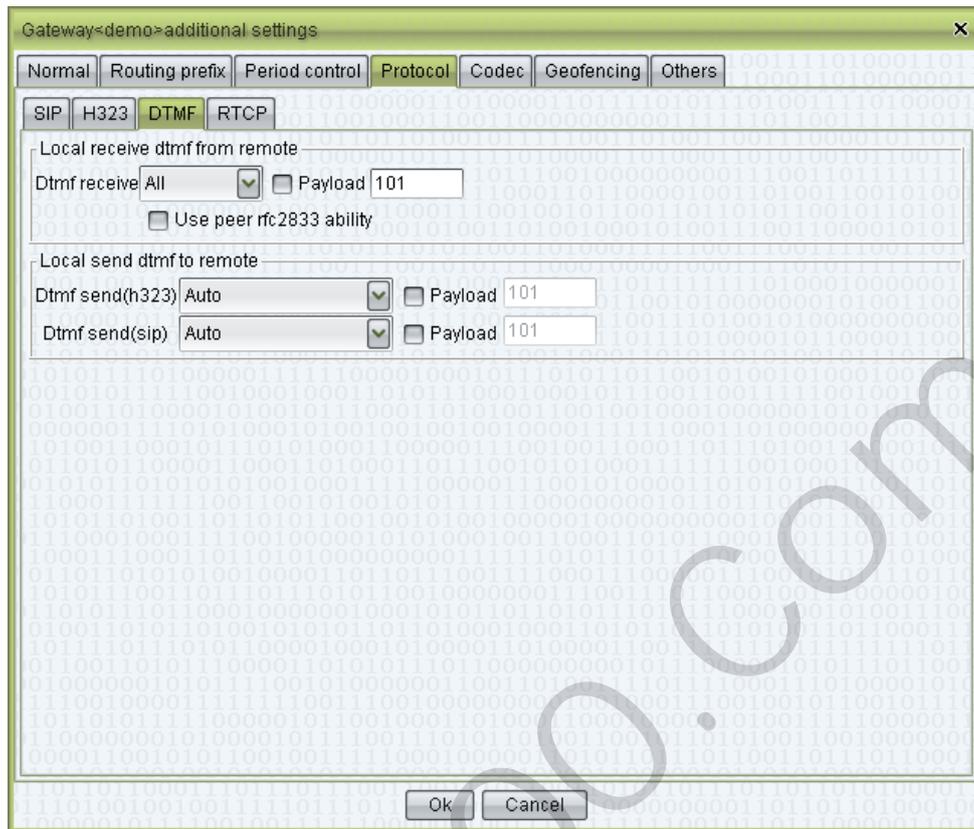
If "Switch gateway until connect" is on, this setting is invalid.

- Q931 NumberingPlan: see H323's RFC.

- Q931 NumberType: see H323's RFC.
- Q931 PresentationIndicator: see H323's RFC.
- Q931 ScreeningIndicator: see H323's RFC.
- G729 negotiation mode:
 - Auto: keep original G729 codec.
 - G729: treat G729a or G729 as G729.
 - G729a: treat G729 or G729a as G729a.
 - G729&G729a: treat G729 or G729a as G729 and G729a.
- FastStart: check to enable.
- H245Tunneling: check to enable.
- H245InSetup: check to enable.
- Allow t38: check to enable.

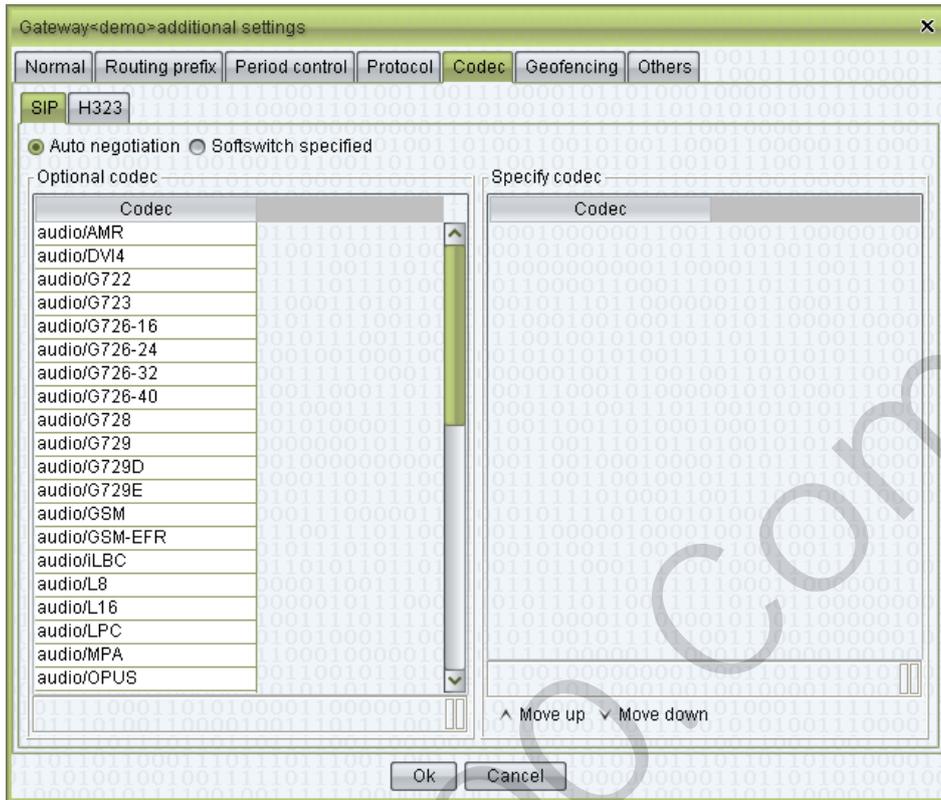
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Additional settings > Protocol > DTMF



- Dtmf receive: specifies the way by which DTMF signals are received. The <All> option is recommended, which asks the system to accept all kinds of DTMFs. Once a certain kind of DTMF is received, this channel will accept the same kind of DTMFs only, thus effectively avoiding duplicate receptions.
- Use peer rfc 2833 ability: rfc 2833 ability is determined by peer ability.
 - After checking, use the rfc2833 support capability of the opposite end (caller), otherwise, the vos3000 declares to support rfc2833 capability.
- Payload: the payload value in RTP, for the DTMF of the RFC2833 mode.
 - For example, if the payload is 97, then the payload value of rfc2833 message must be 97.
- Dtmf send(h323): it is set to “Auto” by default, indicating that the system would determine the best way to send DTMFs based on the receiver’s capacity. If the receiver provides no capacity set, the system will send according to the default mode. The RFC2833 mode can only be specified for media proxy.
- Dtmf send(sip): it is set to “Auto” by default. The details are the same as those of “Dtmf send(h323)” described above.

Additional settings > Codec > H323/SIP



- Auto negotiation: determined by caller and callee.
- Softswitch specified: codec can only be sent.

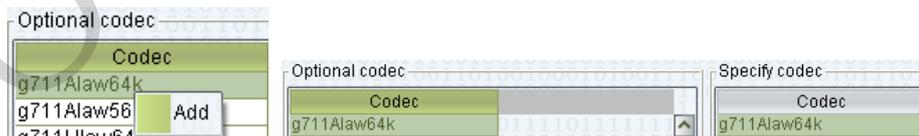
TIP

Select one codec, and then double click to add or delete.
Select some codecs, and then right click to add or delete.

NOTE

For specified codec, "Softswitch specified" is needed.

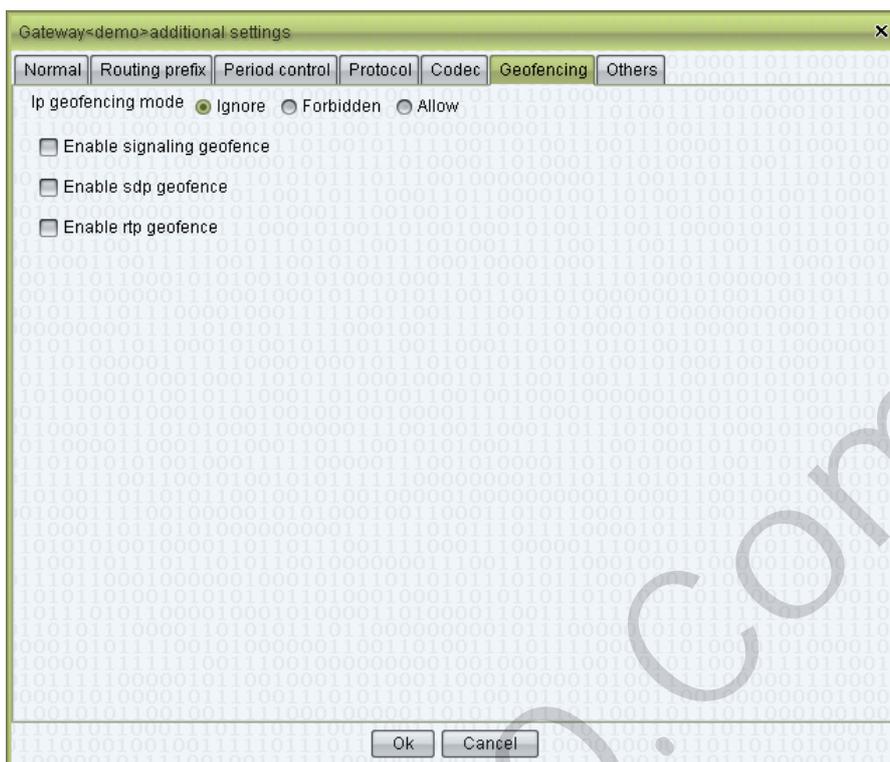
- Add process



- Deletion process



Additional settings > Geofencing



- Ip geofencing mode:
 - Ignore: this feature is not enabled.
 - Forbidden: IP in geographical fence prohibited.
 - Allow: only checked IP within geofence is allowed.
- Enable signaling geofence: detect the signaling address sent by the opposite end.
- Enable sdp geofence: detect the corresponding SDP address after c=INIP4 carried by SDP field in signaling.
- Enable rtp geofence: detect the RTP address sent by the peer in the actual call.

 **NOTE**

The IP address list is configured in "Operation management-Geofencing". See " Operation management-Geofencing " for details.

Additional settings > Others

The screenshot shows a dialog box titled "Gateway<demo>additional settings" with a close button (X) in the top right corner. The dialog has several tabs: "Normal", "Routing prefix", "Period control", "Routing settings", "Protocol", "Codec", "Geofencing", and "Others" (which is currently selected). The "Others" tab contains the following settings:

- Number limit:** Two text input fields for "Caller number allow length" and "Callee number allow length".
- Number restriction on callee:** A dropdown menu set to "None" and a checkbox for "Callee transform".
- Enable phone display number:** A checked checkbox.
- Set to protect route:** An unchecked checkbox.
- Enable dynamic blacklist in standalone mode:** An unchecked checkbox.
- Enable bilateral reconciliation:** An unchecked checkbox.
- Do not disabled when timeout:** An unchecked checkbox.
- Rtp header serialization:** A checked checkbox.
- Passthrough rtp loss rate:** A checked checkbox.
- timestamp:** A checked checkbox.
- Enable caller number pool:** An unchecked checkbox with a dropdown menu set to "Multiplexes 1".
- Enable axb A number group:** An unchecked checkbox with a dropdown menu.
- Axb interface management:** A dropdown menu.
- Callee dial plan:** A text input field.
- Routing clearing account id:** A text input field.
- Enable forwarding signal caller pool:** An unchecked checkbox with a dropdown menu.
- Rate limit:** A dropdown menu set to "Max", followed by "calls every 1000 ms".
- Signaling tracing:** A section containing:
 - Call tracing:** An unchecked checkbox.
 - Register tracing End time:** A text input field set to "2021-12-19 00:00:00" with a dropdown menu.

At the bottom of the dialog are "Ok" and "Cancel" buttons.

- Number length limit:
 - Caller number allowable length: the lengths of the caller numbers allowed to pass through the gateway (e.g. fill in "11, 14" to allow numbers of 11 digits or 14 digits only).
 - Callee number allowable length: the lengths of the called numbers allowed passing through the gateway.



NOTE

Left blank to allow numbers of all length to pass through, and fill in “0” to allow no numbers to pass through.

- Number restriction on callee:
 - None: no restriction.
 - Phone number: allow call platform's phone numbers only.
 - Other number: allow call numbers except platform's phone number.
- Callee transform: use number in "Number Transformation" table to replace callee ID.
- Enable phone display number: when caller is phone, check to use phone's display number, uncheck to use phone number.



NOTE

Gateway's dial plan will still be used.

- Http call status notification: send the call status to the HTTP server.
- Set to protect route: if set to protect route, this gateway will not be used to sort with other gateways. Between protect routes, sort order as normal. If normal gateways are not connected, the call will switch to protect routes. Protect route enable time can be set, within the time call is not connected, will be send to protect routes.
- Enable dynamic blacklist in standalone mode: when the dynamic blacklist run mode is standalone, monitor blacklist dynamically on the gateway.
- Enable bilateral reconciliation: VOS will check the amount deviation of customer and vendor automatically.
- Passthrough rtp lost rate: transparent media message to peer platform.
- Do not disabled when timeout: routing gateway still can be used after timeout.
- Enable caller number pool: use number in pool as caller.
- Multiplexes: the number of repeated uses of each number in the calling number pool is the maximum concurrency limit.
- Enable forwarding signal caller pool: use number in pool as caller.



NOTE

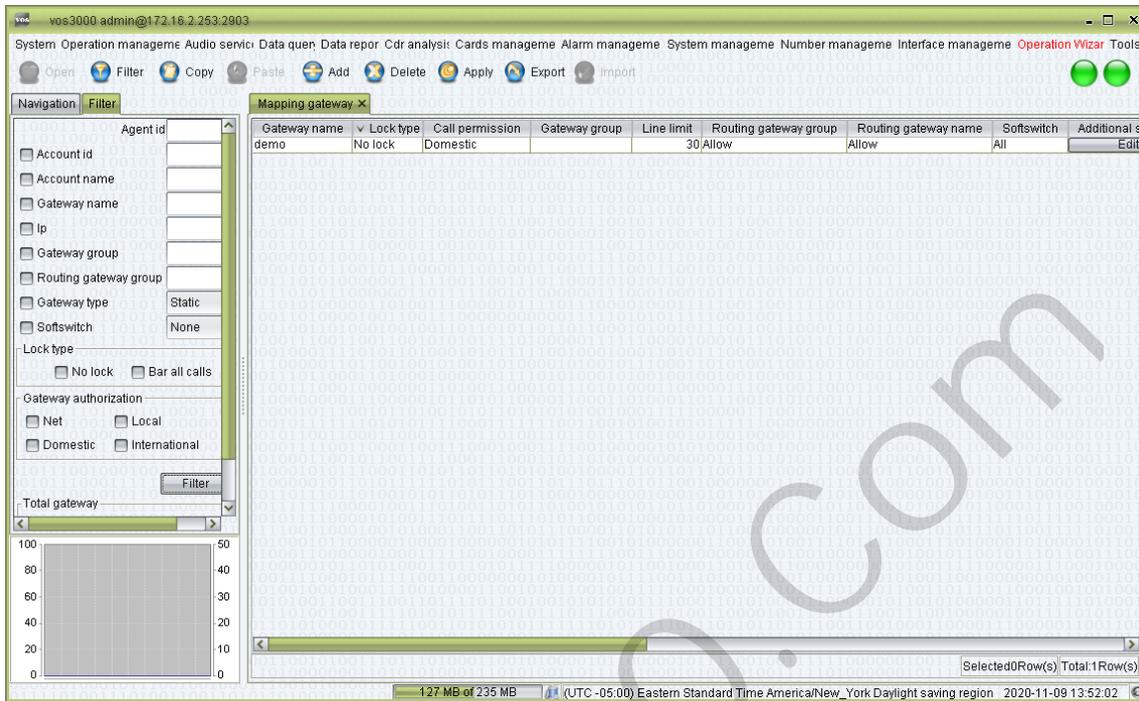
- Number sequence is decided by <FORWARD_SIGNAL_REWRITE_SEQUENCE> setting in config file softswitch.conf
- 0: random
- 1: poll
- Default is random
- Rate limit: Restrict the number of call-establish signaling in a certain period.
- Signaling tracing: Setting this gateway, call tracing, register tracing, end time. This setting will not affected by the system <Debug trace> setting.

Other Operations

- Double-click the content of “Routing clearing account name” to open the account management page for this account.

2.5.1.2 Mapping Gateway

This function is used to manage mapping gateway.



How to Start

- Double-click “Navigation > Operation management > Gateway operation > Mapping gateway”

Table Items

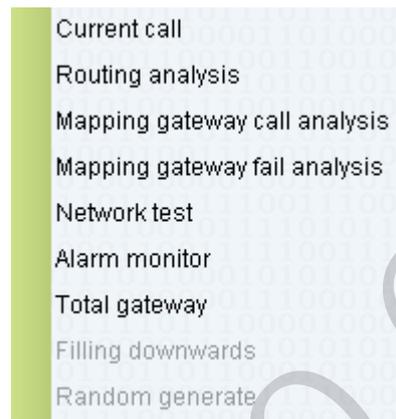
- Gateway name: see the descriptions in “Routing gateway”.
- Lock type: see the descriptions in “Routing gateway”.
- Call permission: see the descriptions in “Phone management”.
- Gateway group: set the gateway’s group, used for control several gateways’ total currency.
- Line limit: the number of concurrent calls allowed by the gateway.
- Routing gateway group: allow or forbidden.
- Routing gateway name: set mapping gateway allow/forbidden to try those routing gateways.
- Softswitch: see the descriptions in “Routing gateway”.
- Additional settings
- Ip: see the descriptions in “Additional settings”.
- Account id: the number of the billing account for this mapping gateway.
- Account name: the name of the billing account for this mapping gateway.
- Configuration password: see the descriptions in “Routing gateway”.
- Self service password: see the descriptions in “Routing gateway”.
- Priority: for static gateway with the same IP, use this priority to match.

- Caller black/white list group: see the descriptions in “Routing gateway”.
- Callee black/white list group: see the descriptions in “Routing gateway”.
- Memo: comments on this gateway.

Other Operations

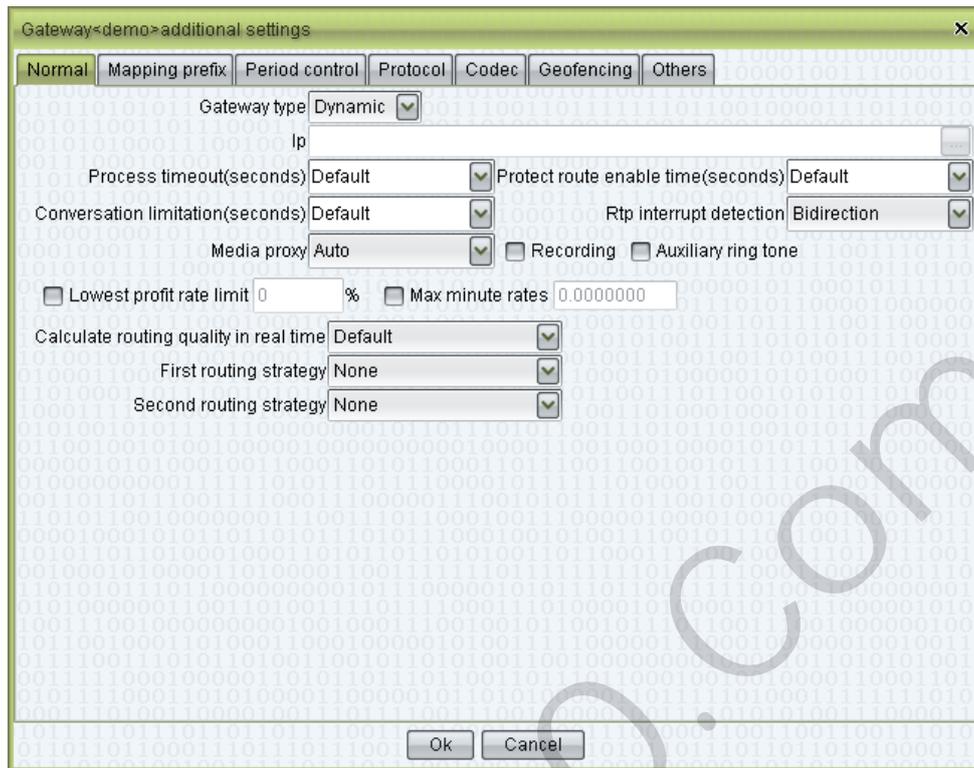
- Double-click the ”Account name”, can directly enter to the manage interface of this account.

Right-Click Menu



- Same as “Routing gateway”.
- Routing analysis: open “Routing analysis” table.
- Mapping gateway call analysis: as “Routing gateway”.
- Mapping gateway fail analysis: as “Routing gateway”.
- Network test: as “Routing gateway”.
- Alarm monitor: as “Routing gateway”.
- Total gateway: as “Routing gateway”.
- Filling downwards: as “Routing gateway”.
- Random generate: as “Routing gateway”.

Additional settings > Normal



- Gateway type:
 - Dynamic: registration is required.
 - Static: gateway mapping is achieved directly through IP addresses.
- Ip: IP addresses of mapping gateways. Multiple addresses and signaling port can be specified, separated by commas.
- Process timeout(seconds): the maximum time waited after the call has reached the gateway. If the connection has not been establish within the time limit, the system server will send a reject signal to the mapping gateway.
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_MAPPINGGATEWAYTIMEOUT”.
 - None: no limitation.
- Protect route enable time(seconds):
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_TRY_PROTECT_ROUTE_DELAY”.
 - None: no limitation.
- Conversation limitation(seconds):
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_MAXCALLDURATION”.
 - None: no limitation.
- RTP interrupt detection: see the descriptions in “Routing management”.
- Media proxy: see the descriptions in “Routing gateway”.
- Recording: need to purchase recording module.

- Auxiliary ring tone: when media forwarding is on, if the called RTP is not received within the specified time (“auxiliary ring tone activation delay”) after receiving the called SDP, the vos3000 plays the auxiliary ring tone through the local RTP. After receiving the called RTP, the called RTP is transmitted.



NOTE

- Auxiliary ring tone activation delay: determined by “Softswitch Management > Additional set > system parameter > SS_AUXILIARY_RING_TONE_ACTIVATION_DELAY” parameter.
- Lowest profit rate limit: this gateway will be locked if profit below this value.

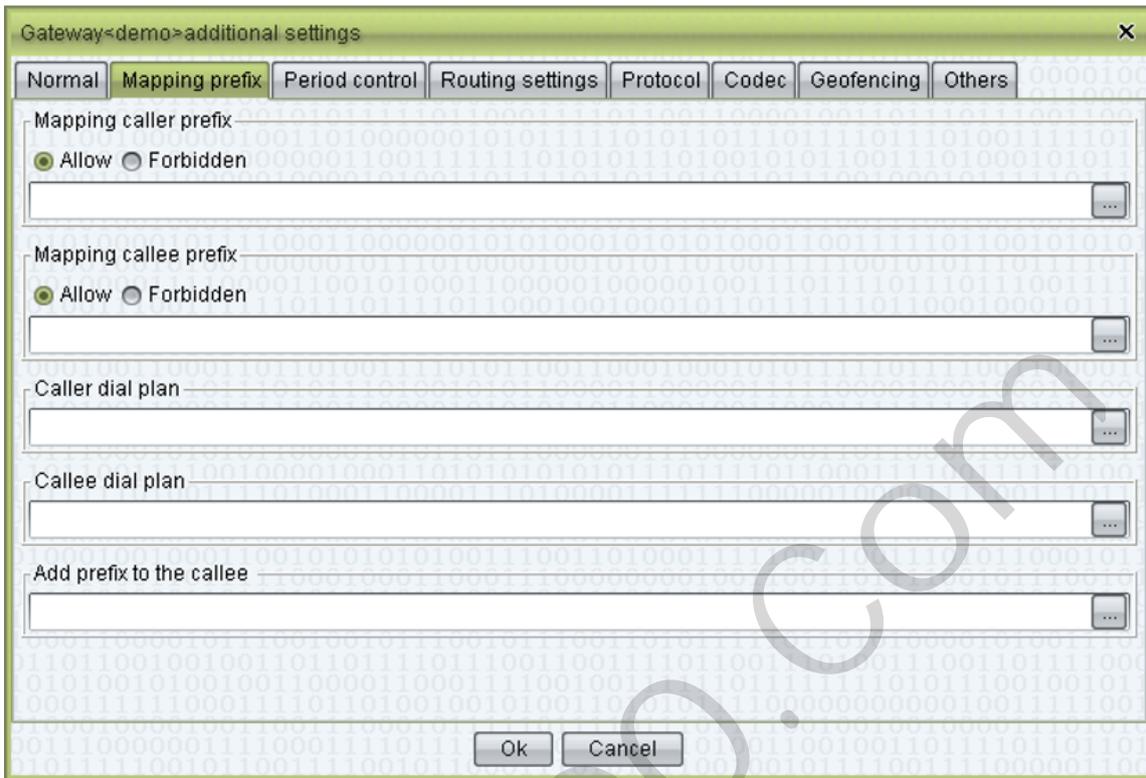


NOTE

- Only the basic rate of the account is used as the calculation basis, excluding the package of the account.
- Max minute rates: max minute rate allowed for callee number.
- Calculate routing quality in real time: as “Routing gateway”.
- First routing strategy:
 - None: System default.
 - Asr: Sort by ASR.
 - Lowest rate per second: sort by rate per second.
- Second routing strategy: see “First routing strategy”.

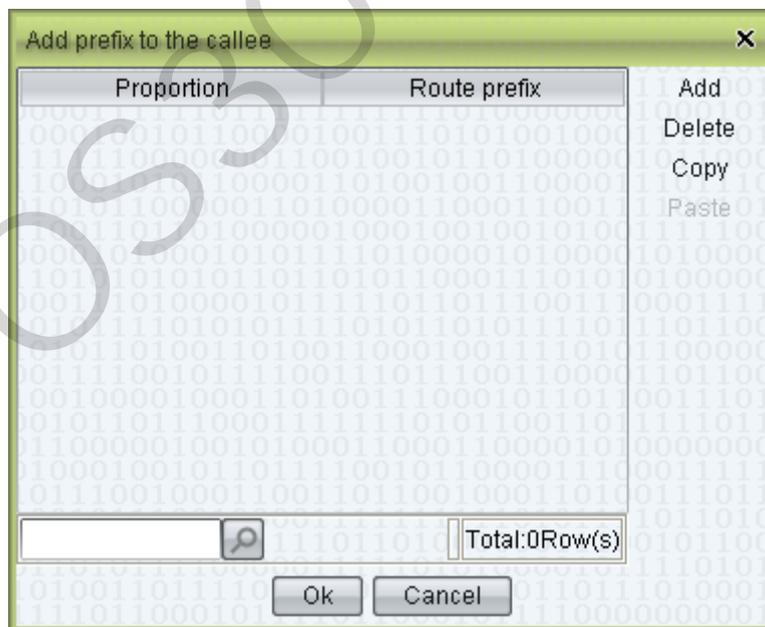
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Additional settings > Mapping prefix



See the descriptions in “Routing gateway”.

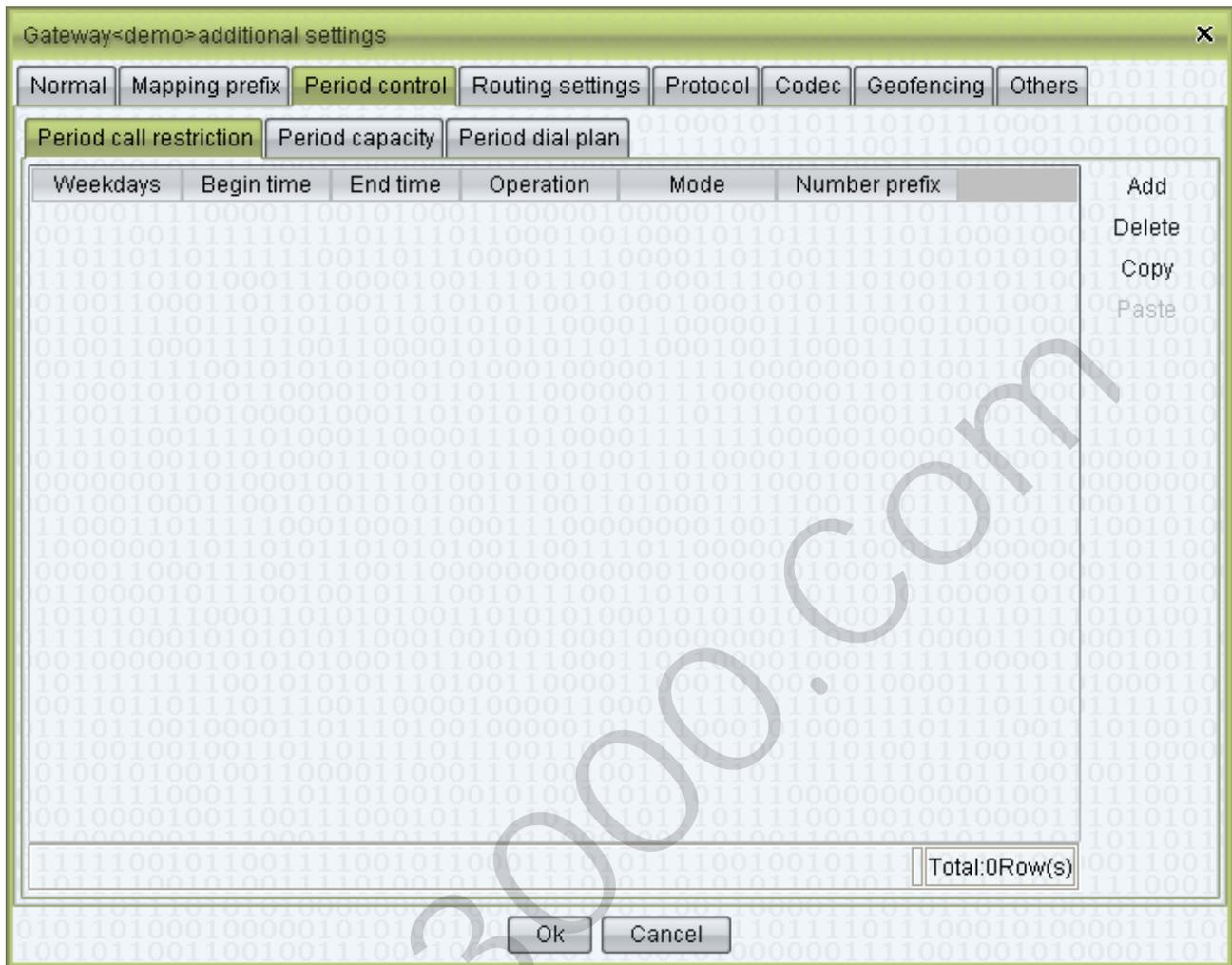
- Add prefix to the callee: add prefix for callee number by proportion.



NOTE

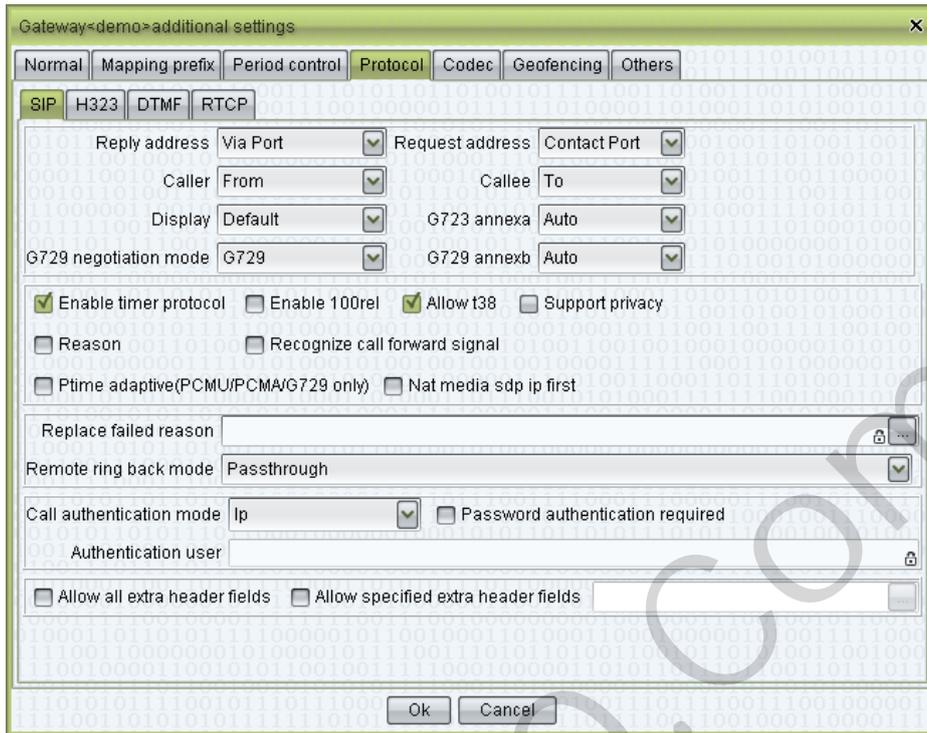
- When “callee dial plan” and “add prefix to callee” are present rules for same callee number at the same time, callee will be applied by rule of “callee dial plan” at first, then rewritten number will be added prefix by proportion.

Additional settings > Period control



See the descriptions in “Routing gateway”.

Additional settings > Protocol > SIP



See the descriptions in “Routing gateway”.

- Caller: get caller number from which field of signal.
 - From: get caller number from “From” of signal.
 - Remote-Party-ID: get caller number from “Remote-Party-ID” of signal.
 - Display: get caller number from “Display” of signal.
- Callee: get callee number from which field of signal.
 - To: get callee number from “To” of signal.
 - Request-Line: get callee number from “Request-Line” of signal.
- Support Privacy: pass through mapping gateway private domain.
- Recongnize call forward signal: if mapping gateway send calls in forward signal format, call with forward format will be identified as forward call.
- Replace failed reason: specify the error message sent to the mapping gateway when the call cannot be established, can define their replacement rules according to different termination reason.

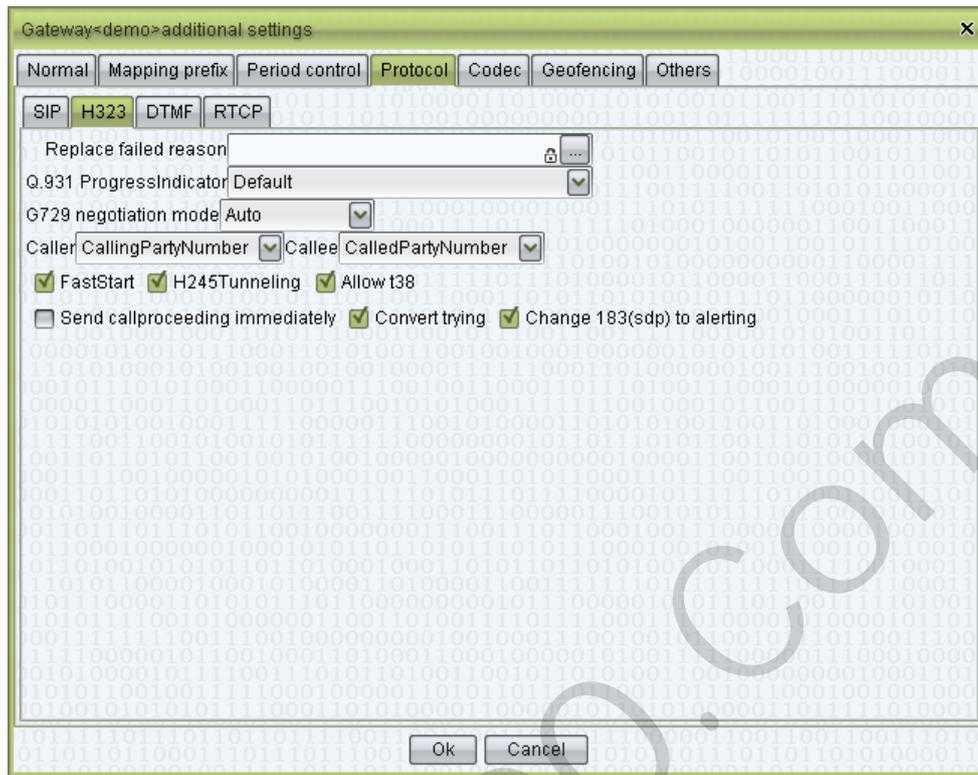


- Remote ring back mode:
 - Passthrough: according to routing gateway.
 - 183 Session Progress + SDP: when callee ringing, send 183 Session Progress + SDP to mapping gateway.

- 180 Alerting + SDP: when callee ringing, send 180 Alerting + SDP to mapping gateway.
- Call authentication mode:
 - IP: verify IP Address.
 - IP Address and Port: verify IP Address and Port.
 - Password authentication required: using password authentication method.
- Allow all extra header fields: SIP header transparent, allowing all additional header domains.
- Allow specified extra header fields: Additional header fields can be added and specified independently.

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Additional settings > Protocol > H323

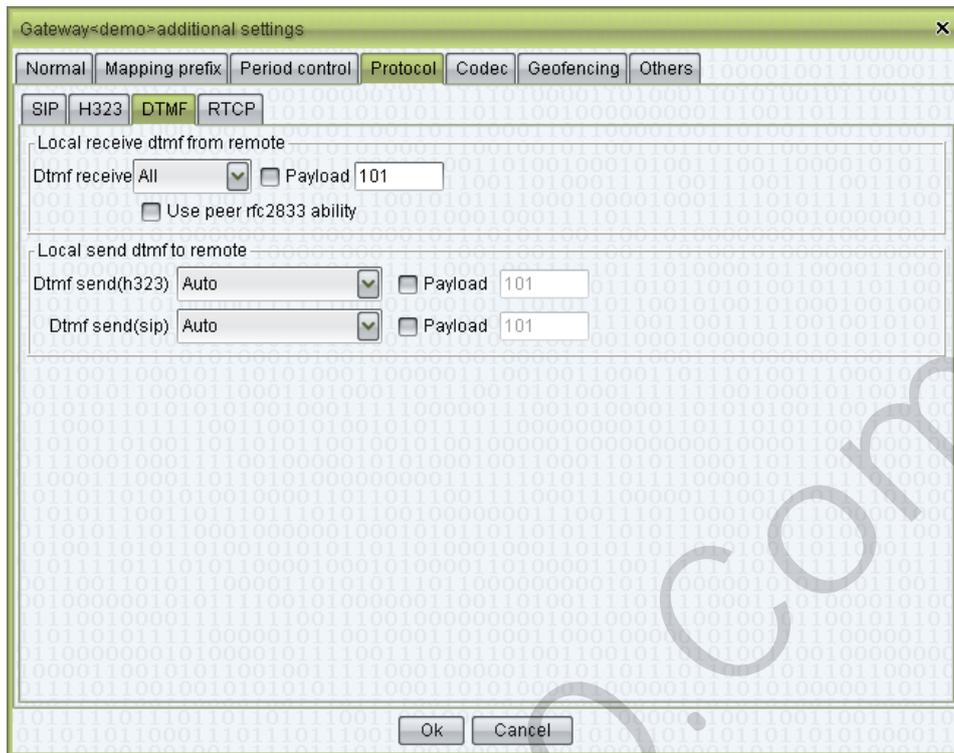


- Replace failed reason: specify the error message sent to the mapping gateway when the call cannot be established, can define their replacement rules according to different termination reason.



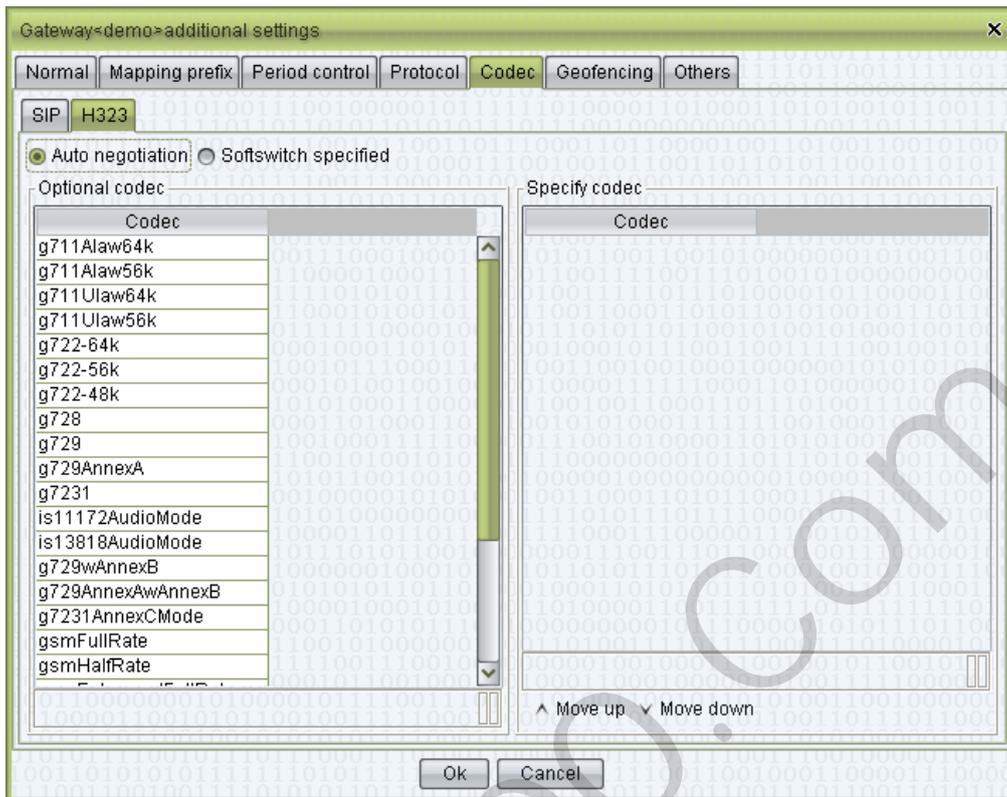
- Q.931 ProgressIndicator: see the standard reference for H323. See the descriptions in “Phone management” for details about signaling checkboxes.
- G729 negotiation mode
- Caller: CallingPartyNumber/SourceAddress/Display.
- Callee: CalledPartyNumber/DestinationAddress.
- FastStart: check to enable.
- H245Tunneling: check to enable.
- Allow t38: check to include T38.
- Send callproceeding immediately: check to send callproceeding when receives setup.
- Convert trying: check, if caller uses H323 and callee uses SIP, when callee returns Trying, VOS will send callproceeding to caller.
- Change 183(sdp) to alerting: check, if caller uses H323 and callee uses SIP, when callee returns 183 with SDP, VOS will send alerting to caller; Uncheck, VOS will send callproceeding to caller.

Additional settings > Protocol > DTMF



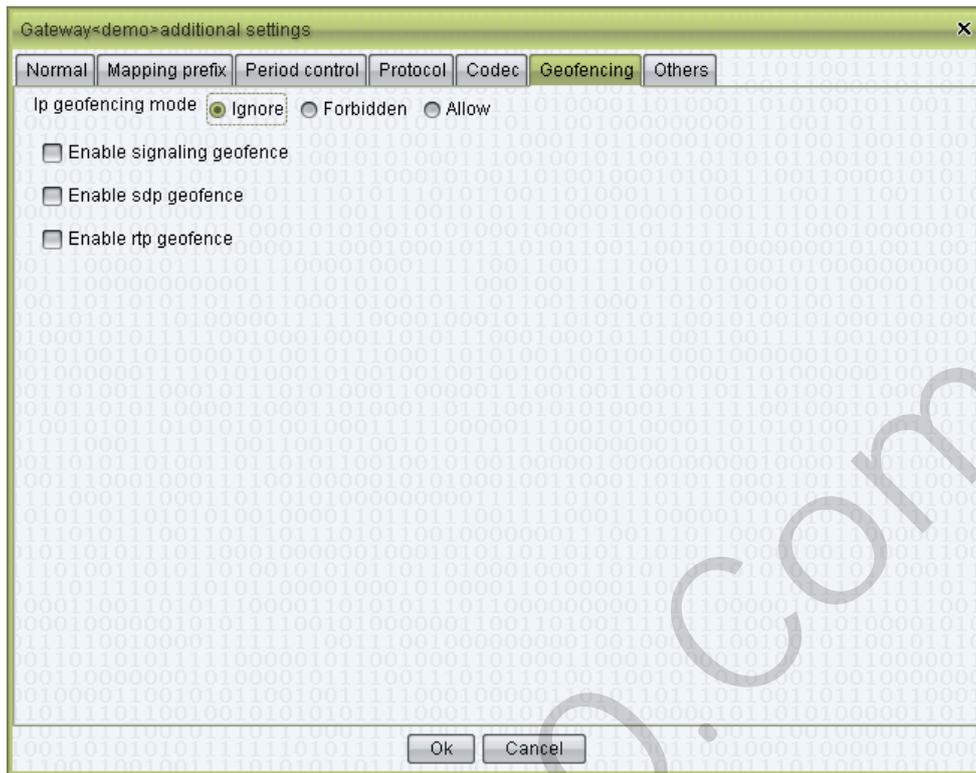
See the descriptions in “Routing gateway”.

Additional settings > Codec



See the descriptions in "Routing gateway".

Additional settings > Geofencing



See the descriptions in “Routing gateway”.

Additional settings > Others

Gateway<demo>additional settings

Normal Mapping prefix Period control Routing settings Protocol Codec Geofencing Others

Number limit

Caller number allow length

Callee number allow length

Caller transform

Allow phone number billing Enable phone settings

Audio prompts for can't connect Language name

Not recorded no hold time cdr

Enable dynamic blacklist in standalone mode

Allow bilateral reconciliation

Rtp header serialization Passthrough rtp loss rate timestamp

Number restriction on callee

Enable forwarding signal caller pool

Rate limit calls every ms

Signaling tracing

Call tracing Register tracing End time

Ok Cancel

See the descriptions in “Routing gateway”.

- Caller number allow length: allowed length of caller number (eg. fill in “11, 14” ,means the gateway only allow 11-digit caller number and 14-digit caller number to pass through).
- Callee number allow length: allowed length of callee number.
- Caller transform: use number in "Number Transformation" table to replace caller ID.
- Http call status notification: send the call status to the HTTP server.
- Allow phone number billing: if caller is platform’s phone number, use account that phone number belongs to billing.
- Enable phone settings: when the calling number is the number in "Phone management", the configuration information is used for subsequent processing
- Audio prompts for can’t connect: if calls fail to connect, system will broadcast the failure reason.
- Language name: Chinese/English.
- Not recorded no hold time cdr: system will not record the cdr with no conversation time.
- Allow bilateral reconciliation: VOS will check the amount deviation of customer and vendor automatically.

- Passthrough rtp loss rate: transparent media message to peer platform.
- Number restriction on callee:
 - None: no restriction.
 - Phone number: only platform's phone number is allowed.
 - Other number: only number not platform's number is allowed.
- Enable forwarding signal caller pool: use number in pool as caller.



NOTE

In American, location routing number (LRN) is saved in signal control point (SCP). LRN is used for number portability.

LRN server can be set in "Softswitch management > Additional settings > System parameter > SS_LRN_SERVER_IP" and "SS_LRN_SERVER_PORT".

- Eat prefix length: cut the number, then use left to billing.
- Failure action: when query failed, "Reject" or "Continue".
- Routing using number: when set to query, indicates that the route lookup is based on the number returned by the LRN server.
- Interstate billing prefix: add prefix for interstate call.



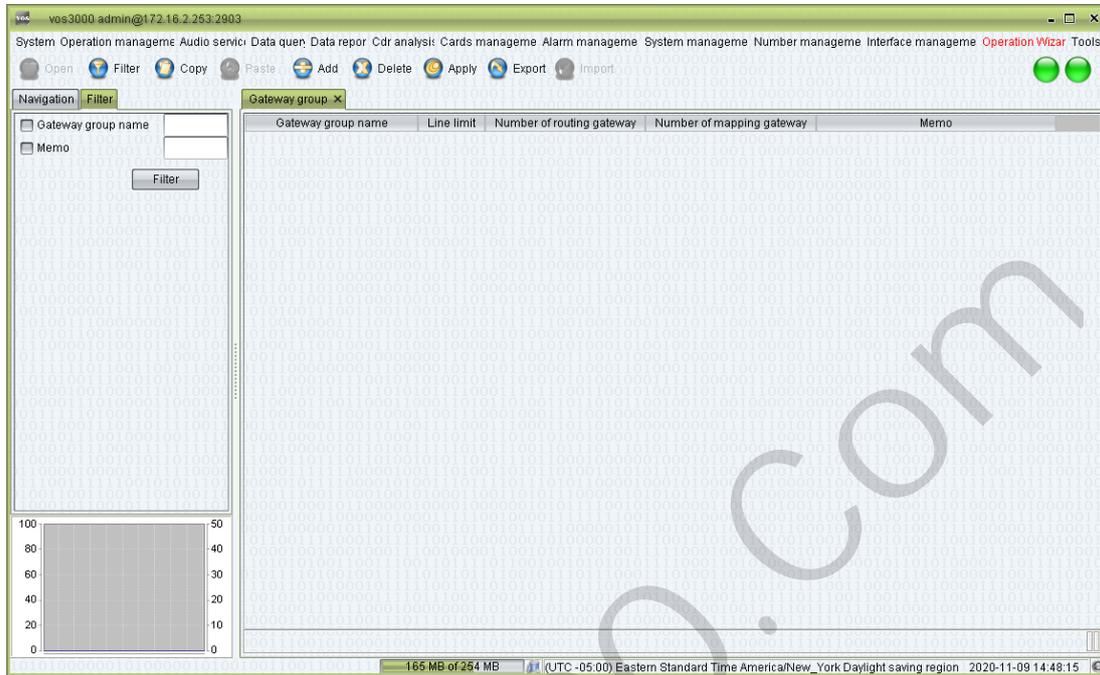
NOTE

Identify of interstate can be referred to "Number Management > Lerg Data", if caller and callee's state are different, the call is interstate.

- Undetermined billing prefix: add prefix for the calls which cannot be recognized interstate.
- Rate limit: Restrict the number of call-request in a certain period.
- Signaling tracing: Setting this gateway, call tracing, register tracing, end time. This setting without restriction of the system <Debug trace> setting.

2.5.1.3 Gateway Group

This function is used to define gateway groups, which is used to limit gateways during routing.



How to Start

- Double-click “Navigation > Operation management > Gateway operation > Gateway group”

Table Items

- Gateway group name: the name of gateway group.
- Line limit: total capacity of the gateway group.
 - None: means lines of all gateways included in the group are all available.
 - Use gateway setting: means gateway’s line limit setting will be used.
- Number of routing gateway: double-click to manage.
- Number of mapping gateway: double-click to manage.
- Memo: additional comments.

Other operations

- double-click the number of routing gateways to directly enter the "routing gateway" of the gateway group.
- double-click the number of mapping gateways to directly enter the "mapping gateway" of the gateway group.



NOTE

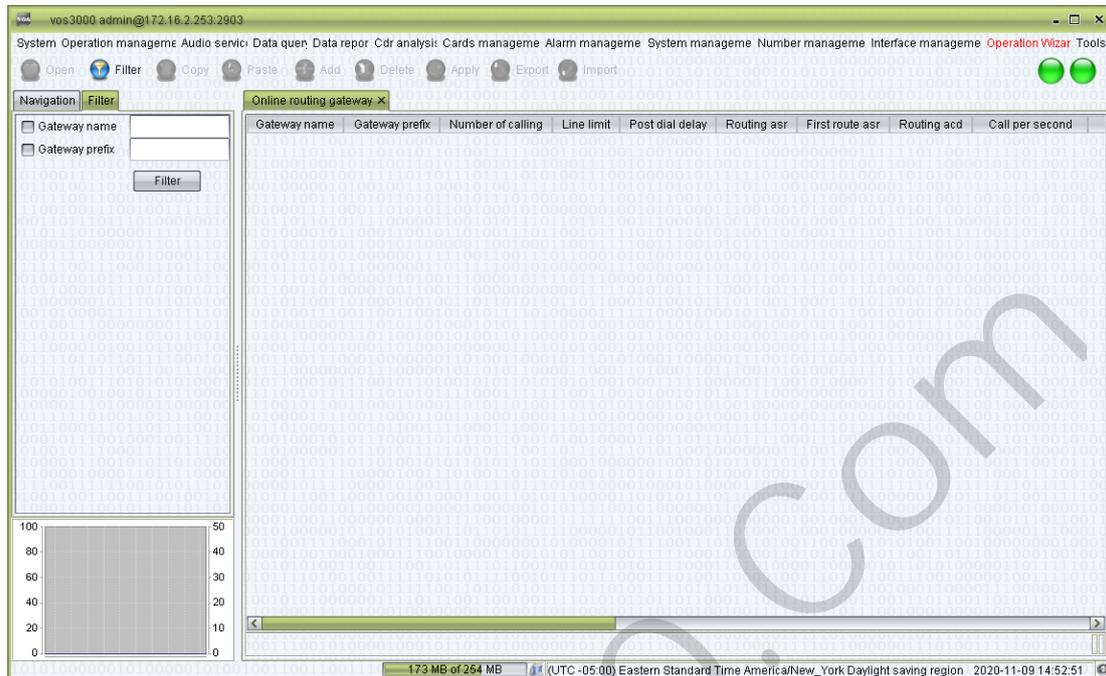
If phone or mapping gateway's Allow gateway groups is deleted totally, means forbidden all routing gateways.

If phone or mapping gateway's Forbidden gateway groups is deleted totally, means allow all routing gateways.

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2.5.1.4 Online Routing Gateway

This function is used to query online routing gateway.



How to Start

- Double-click “Navigation > Operation management > Gateway operation > Online routing gateway”

Table Items

- Gateway name: the device id of gateway.
- Gateway prefix: the prefix of gateway.
- Number of calling: the number of current sessions maintained by the gateway and the total capacity of it.
- Line limit: lines of this routing gateway.
- Routing asr: display current asr if routing gateway open “Real time computing asr”.
- First route asr: the first asr use the routing gateway.
- Routing acd: display current acd if routing gateway open “Real time computing acd”.
- Call per second: display the current call rate when routing gateway <rate limit> is turned on .
- Registered ip: the current IP of the gateway.
- Registration time: the server time of the platform's most recent registration.
- Update time: the time of the most recent confirmation that the platform is online.
- Duration: the time elapsed since the most recent registration (for dynamic gateways). (* There is no “Time elapsed” item for static gateways).
- Encryption type: the type of encryption used by the gateway.
- Register name: if “Gateway type” is “Registration”, then display “Mark” in “Registration management”.

- Disabled time: current remaining disabled duration.
- Call tracing: current call tracing status.
- Register tracing: current register tracing status.
- Local ip: send calls by this IP.
- Softswitch name: the name of the softswitch that the gateway registered.

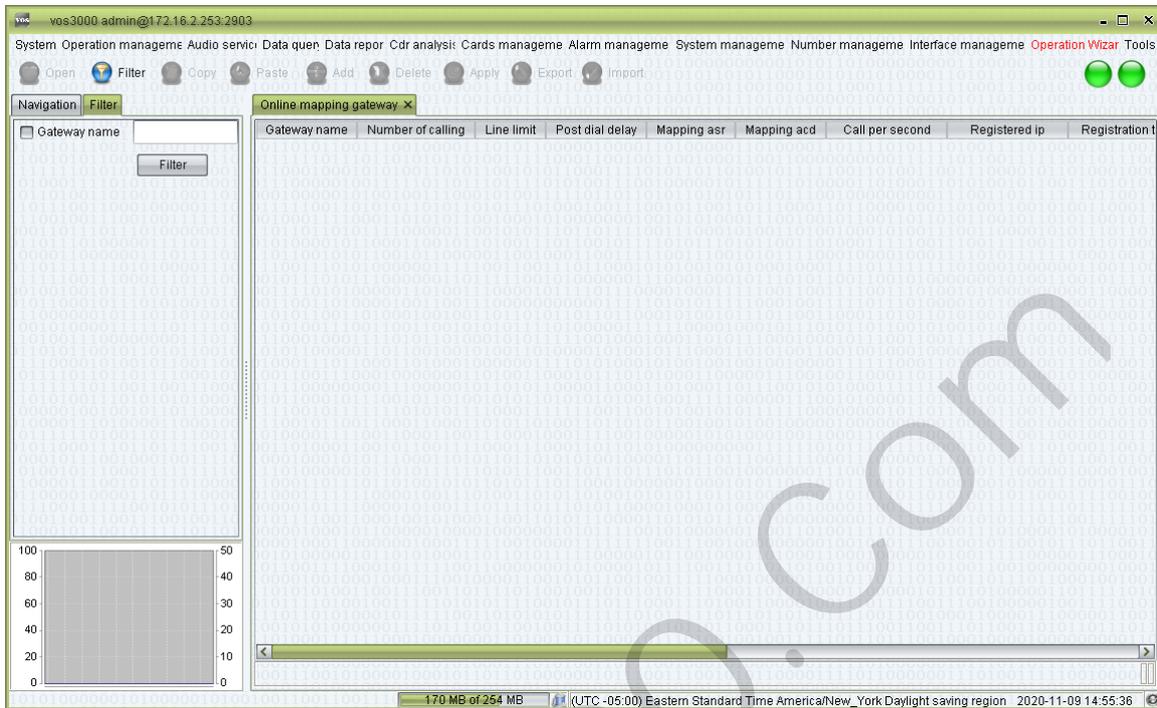
Right-Click Menu

- Current call: open the “Current call” page for this gateway.

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2.5.1.5 Online Mapping Gateway

This function is used to query online mapping gateway.



How to Start

- Double-click “Navigation > Operation management > Gateway operation > Online mapping gateway”

Table Items

- Gateway name: the device ID of gateway.
- Number of calling: the number of current sessions maintained by the gateway and the total capacity of it.
- Line limit: lines of this mapping gateway.
- Mapping asr: display current asr if mapping gateway open “Real time computing asr”.
- Mapping acd: display current asr if mapping gateway open “Real time computing acd”.
- Call per second: display the current call rate when routing gateway <rate limit> is turned on.
- Registered ip: the current IP of the gateway.
- Registration time: the server time of the platform’s most recent registration.
- Update time: the time of the most recent confirmation that the platform is online.
- Duration: the time elapsed since the most recent registration (for dynamic gateways).

NOTE

There is no “Time elapsed” item for static gateways.

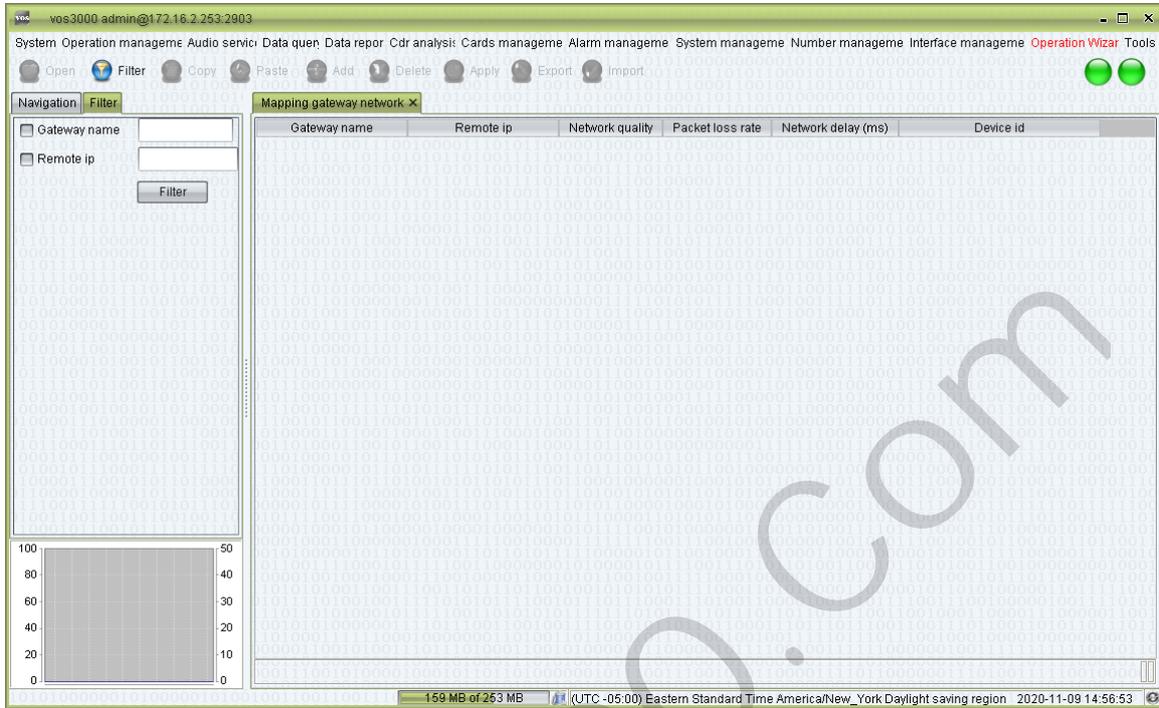
- Encryption type: the type of encryption used by the gateway.
- Call tracing: current call tracing status.
- Register tracing: current register tracing status.

- Local ip: this item is empty, call returns from the original address.
- Softswitch name: the name of the softswitch that the gateway registered.

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2.5.1.6 Mapping Gateway Network

This function is used to view the network status of mapping gateway.



How to Start

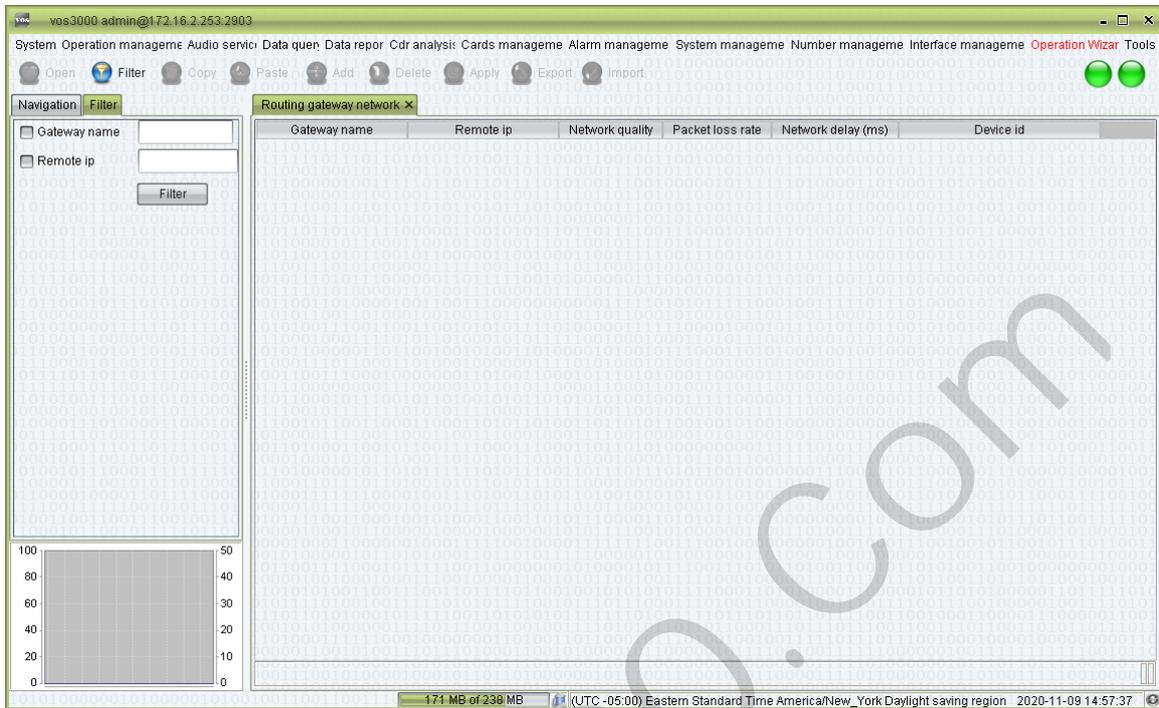
- Double-click “Navigation > Operation management > Gateway operation > Mapping gateway network”

Table Items

- Gateway name: name of mapping gateway.
- Remote ip: IP address.
- Network quality
- Packet loss rate
- Network delay (ms)
- Device id

2.5.1.7 Routing Gateway Network

This function is used to view the network status of routing gateway.



How to Start

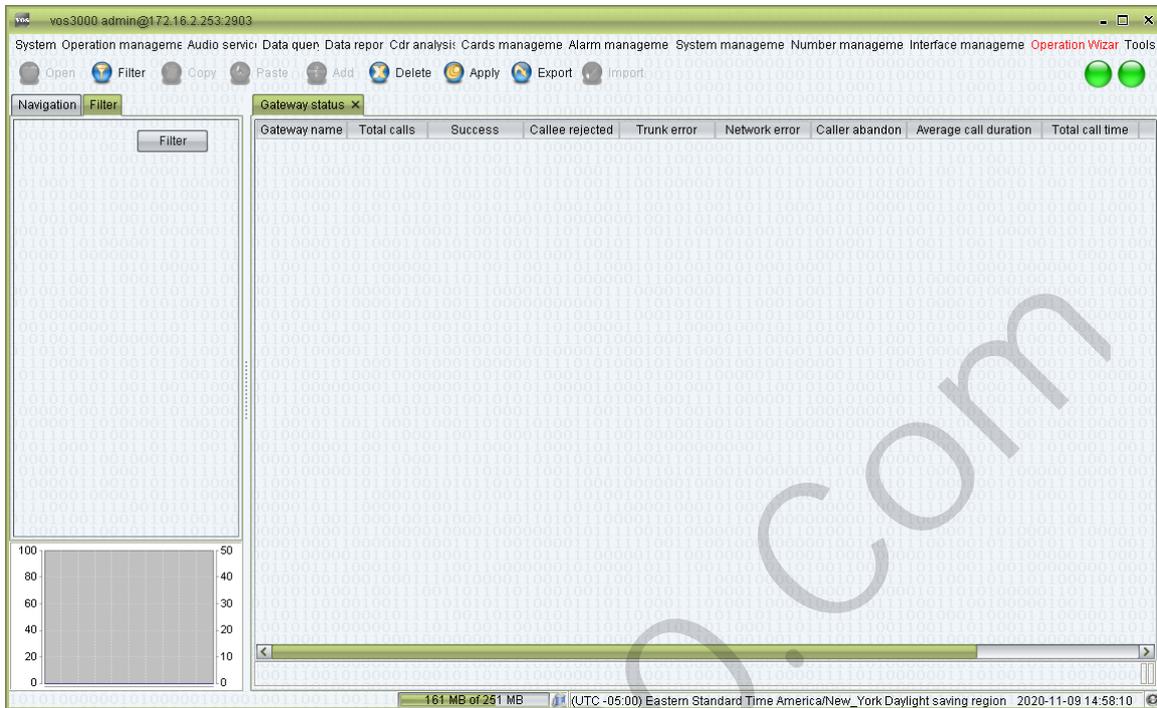
- Double-click “Navigation > Operation management > Gateway operation > Routing gateway network”

Table Items

- Gateway name: name of routing gateway.
- Remote ip: IP address.
- Network quality
- Packet loss rate
- Network delay (ms)
- Device id

2.5.1.8 Gateway Status

This function is query gateway status.



How to Start

- Double-click “Navigation > Operation management > Gateway management > Gateway status”

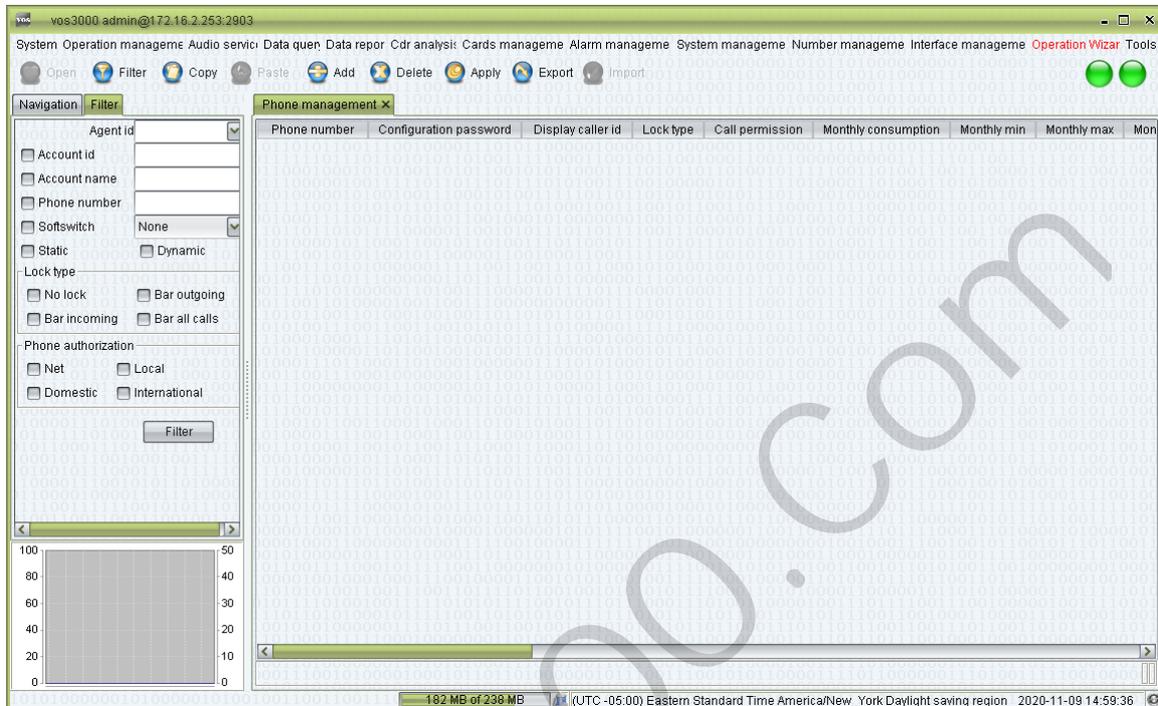
Table Items

- Gateway name
- Total calls
- Success
- Callee rejected
- Trunk error
- Network error
- Caller abandon
- Average talk duration
- Total call time
- Ip
- Starting time

2.5.2 Phone Operation

2.5.2.1 Phone Management

This function is used to manage phone.



How to Start

- Double-click “Navigation > Operation management > Phone operation > Phone management”

Table Items

- Phone number: the number used as caller ID and the called number for the terminal.
- Configuration password: the password used for terminal registration (For H32, will be the H323ID)
- Display caller id: the caller ID shown at the called end.
 - “Remote-Party-ID” to use the number of the original caller.
 - “Display” to use number in “From” field of SIP message.
- Lock type:
 - No lock: none restrictions to the terminal.
 - Bar outgoing: the terminal is not allowed to call out.
 - Bar incoming: the terminal is not allowed to be called.
 - Bar all calls: the terminal is denied from any service.
- Authorization type: when a call is initiated by this number, this type will be compared with the “Rate Type” of the rate. If the “Rate type” of the matching rate has higher precedence than the type specified here, the call will be denied.



NOTE

The precedence of authorization is: International call > Domestic call > Local call > Net call.

- Monthly consumption: consumption so far that month.
- Monthly min: consumption at least per month.



NOTE

At the beginning of the month, system will calculate the cost of last month. If the phone's consumption less than the value, system will take off the difference.

- Monthly max: consumption at most per month.
- Monthly service fee: rent fee.



NOTE

Take off at the beginning of month.

- Billing rate: private rate of the phone.
- Supplementary service
- Advanced configuration



TIP

Dial plans can be used to implement dial local number without add area code, please refer to “dial plan” chapter.

- Routing gateway group: allow routing gateways.
- Account id: editable. Designating the billing account for this terminal.



NOTE

Rate of this account will be used to bill and perform authorization checks upon calling. The operator that edits this number must have authorizations to manipulate phones of the concerned account.

- Account name: non-editable. When the account number is correctly set, the corresponding name of the account will be shown here.



NOTE

Changing the account number will cause changes to the account and the number segment restrictions of its agent.

Double-click to the account management page for this account.

- DID/DDI: after the phone on line, the other numbers allowed as caller ID or callee number.



NOTE

Number only as signaling transfer, not as billing and reporting datas.

- Softswitch: specify the softswitch used by this phone.
- Reverse charging: if “On”, any call to the number will be charge on this account, instead of the caller.
- Self service password: the password used by users to login from the web and query bills. When left blank, the “Configuration password” will be used.
- Call in limitation: when phone is called, maximum of incoming calls.
- Call out limitation: when phone is calling, maximum of outgoing calls.
- Line limit: the maximum number of channels for this phone, which limits the maximum sum of incoming and outgoing calls processed simultaneously.
- Directory: number of phone's phonebook.
- Directory limit: max number of phone's phonebook.
- Caller black/white list group: see the descriptions in “Routing gateway”.
- Callee black/white list group: see the descriptions in “Routing gateway”.

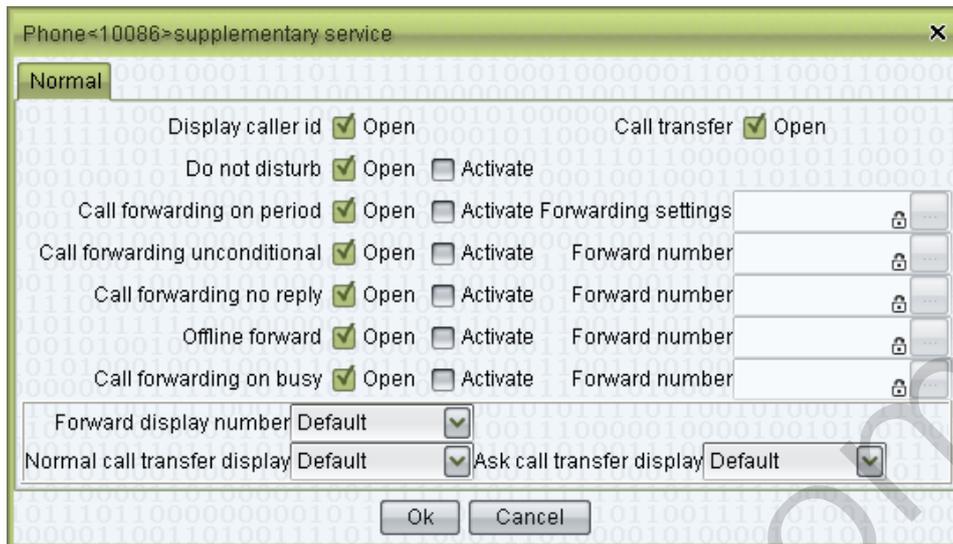
- Memo: descriptions to this phone.

Other operations

- double-click “account name’ to enter the account management of the account

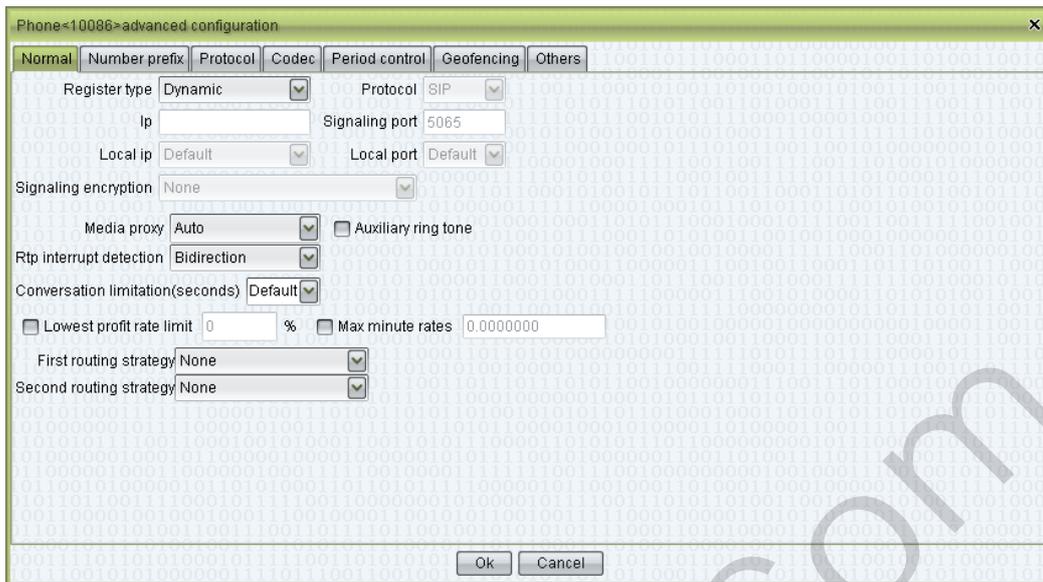
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Supplementary service



- Display caller id: display the caller's ID.
- Call transfer: forward the calls.
- Do not disturb: reject all calls.
- Call forwarding on period: forward calls in specified time period.
- Call forwarding unconditional: forward all calls to specified number.
- Call forwarding no reply: forward calls to specified number when the call is not answered or the phone is out of connection.
- Offline forward: forward calls when phone is not online
- Call forwarding on busy: forward incoming calls when the phone is busy.
- Forward display number:
 - Default: Use “Softswitch management > Additional settings > System parameter > SS_CALL_FORWARD_USING_ORIGINAL_CALLER”.
 - Original caller: display caller’s number.
 - Local number: display the phone’s number.
- Normal call transfer display:
 - Default: Use “Softswitch management > Additional settings > System parameter > SS_CALL_TRANSFER_NORMAL_DISPLAY”.
 - Original caller: display caller’s number.
 - Local number: display the phone’s number.
- Ask call transfer display:
 - Default: Use “Softswitch management > Additional settings > System parameter > SS_CALL_TRANSFER_ASK_DISPLAY”.
 - Original caller: display caller’s number.
 - Local number: display the phone’s number.

Advanced configuration > Normal



See the descriptions in “Routing gateway” or “Mapping gateway”.

- Register type:
 - Static: use IP to access.
 - Dynamic: use register to access.
 - Mapping gateway: use gateway id to access.
- Protocol
- Ip
- Signaling port
- Local ip
- Media proxy
- Rtp interrupt detection
- Conversation limitation(seconds)
- Auxiliary ring tone: when media forwarding is on, if the called RTP is not received within the specified time (“auxiliary ring tone activation delay”) after receiving the called SDP, the vos3000 plays the auxiliary ring tone through the local RTP. After receiving the called RTP, the called RTP is transmitted.

NOTE

Auxiliary ring tone activation delay: determined by “Softswitch Management > Additional set > system parameter > SS_AUXILIARY_RING_TONE_ACTIVATION_DELAY” parameter.

- Try to use routing gateway when phone is offline: when the phone being called is offline, try to find a matching route in the routing gateway.
 - Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_REDIRECTOFFLINEPHONETOGW”.
 - On: try routing gateway.
 - Off: won’t try routing gateway.
- Lowest profit rate limit
- Max minute rates
- First routing strategy

- Second routing strategy

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Advanced configuration > Number prefix

Phone=10086>advanced configuration

Normal Number prefix Routing settings Protocol Codec Period control Geofencing Others

Call in caller prefix
 Allow Forbidden

Call out callee prefix
 Allow Forbidden

Incoming caller dial plan

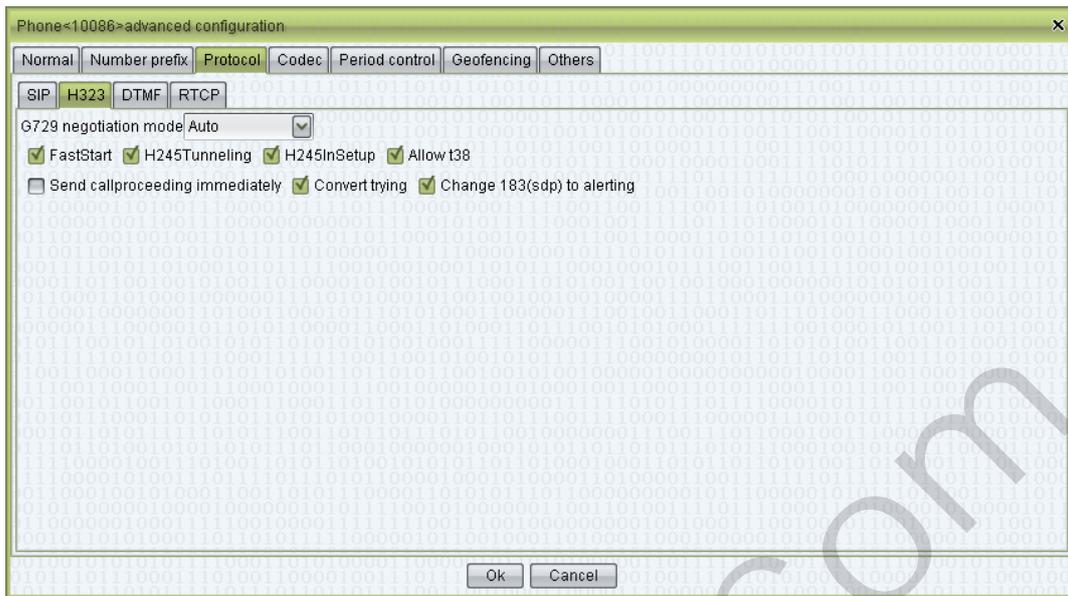
Incoming callee dial plan

Outbound callee dial plan

Ok Cancel

See the descriptions in “Routing gateway”.

Advanced configuration > Protocol > H323



See the descriptions in “Mapping gateway”.

Advanced configuration > Protocol > SIP

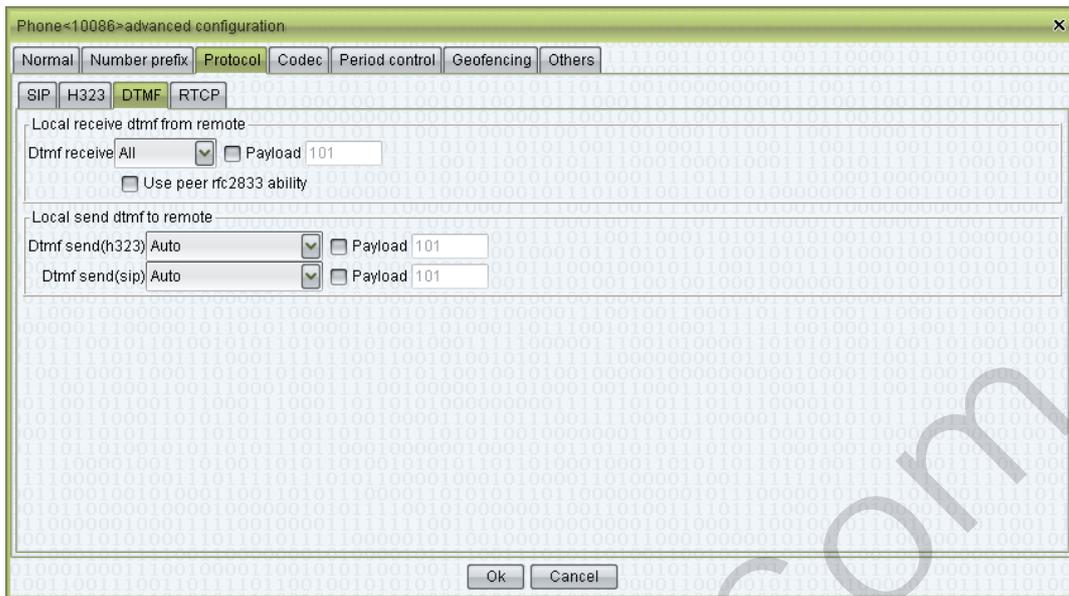
The screenshot shows a configuration window titled "Phone<10086>advanced configuration" with a close button (X) in the top right corner. The window has several tabs: "Normal", "Number prefix", "Protocol" (selected), "Codec", "Period control", "Geofencing", and "Others". Under the "Protocol" tab, there are sub-tabs for "SIP", "H323", "DTMF", and "RTCP". The "SIP" sub-tab is active, showing various configuration options:

- Reply address: Via Port (dropdown)
- Request address: Contact Port (dropdown)
- Display: Default (dropdown)
- G723 annexa: Auto (dropdown)
- G729 negotiation mode: G729 (dropdown)
- G729 annexb: Auto (dropdown)
- Enable timer protocol: (checked)
- Enable 100rel: (unchecked)
- Allow t38: (checked)
- Support privacy: (unchecked)
- Reason: (unchecked)
- Enable local domain name: (unchecked)
- Ptime adaptive(PCMU/PCMA/G729 only): (unchecked)
- New check box: (unchecked)
- Peer number information:
 - Display: Caller (dropdown)
 - Remote-Party-ID screen: None (dropdown)
- Privacy: None (dropdown)
- P-Preferred-Identity: None (dropdown)
- P-Asserted-Identity: None (dropdown)
- Remote ring back mode: Passthrough (dropdown)
- Call authentication mode: Ip (dropdown)
- Password authentication allowed: (checked)
- Password authentication required: (unchecked)
- Trigger on line: (checked)
- Allow all extra header fields: (unchecked)
- Allow specified extra header fields: (unchecked)

At the bottom of the window are "Ok" and "Cancel" buttons.

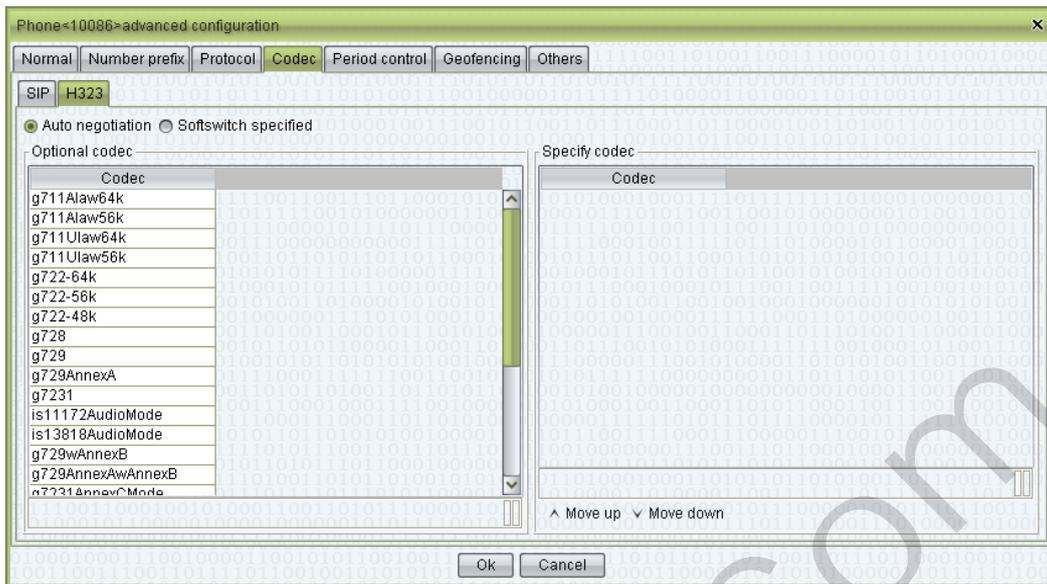
See the descriptions in “Mapping gateway”.

Advanced configuration > Protocol > DTMF



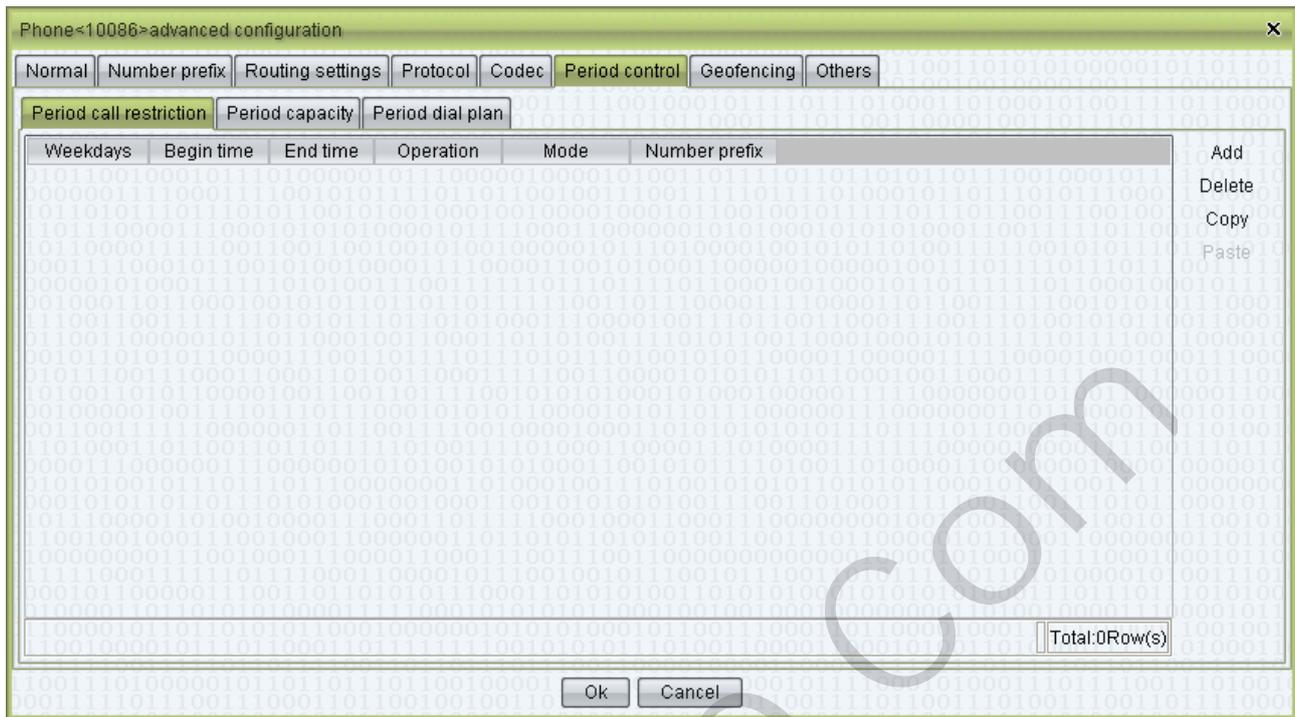
See the descriptions in “Routing gateway”.

Advanced configuration > Protocol > Codec



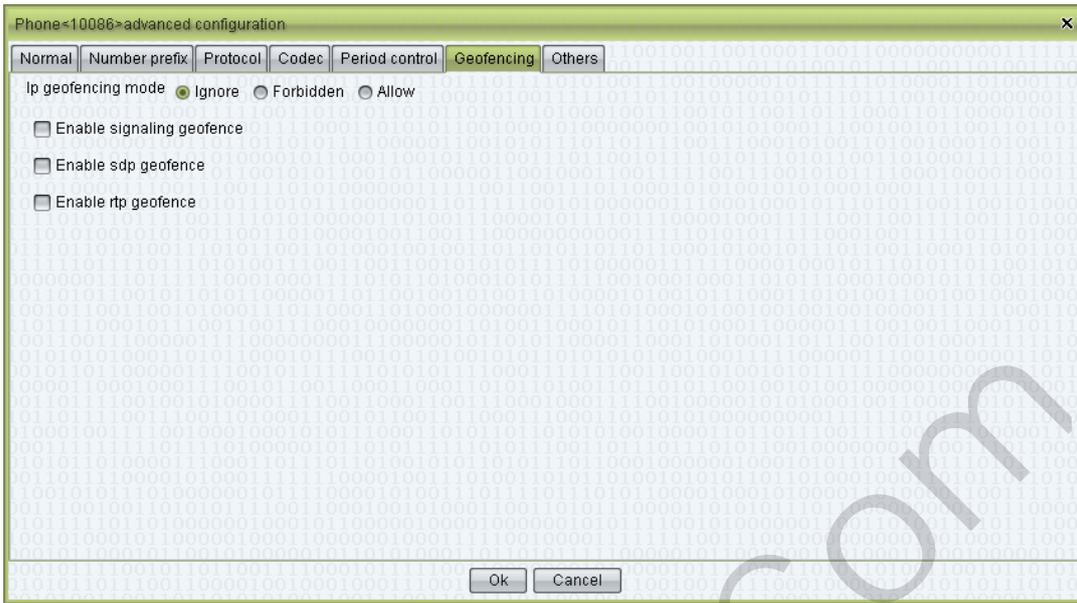
See the descriptions in “Routing gateway”.

Advanced configuration > Period control



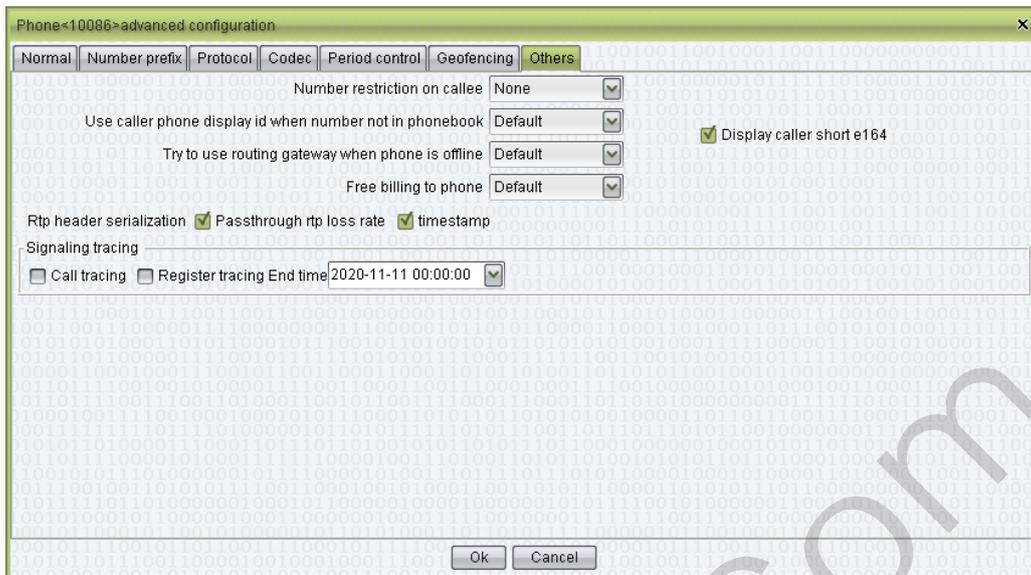
See the descriptions in “Routing gateway”.

Advanced configuration > Geofencing



Audio service is a standalone call service device.

Advanced configuration > Others



See the descriptions in “Routing gateway”.

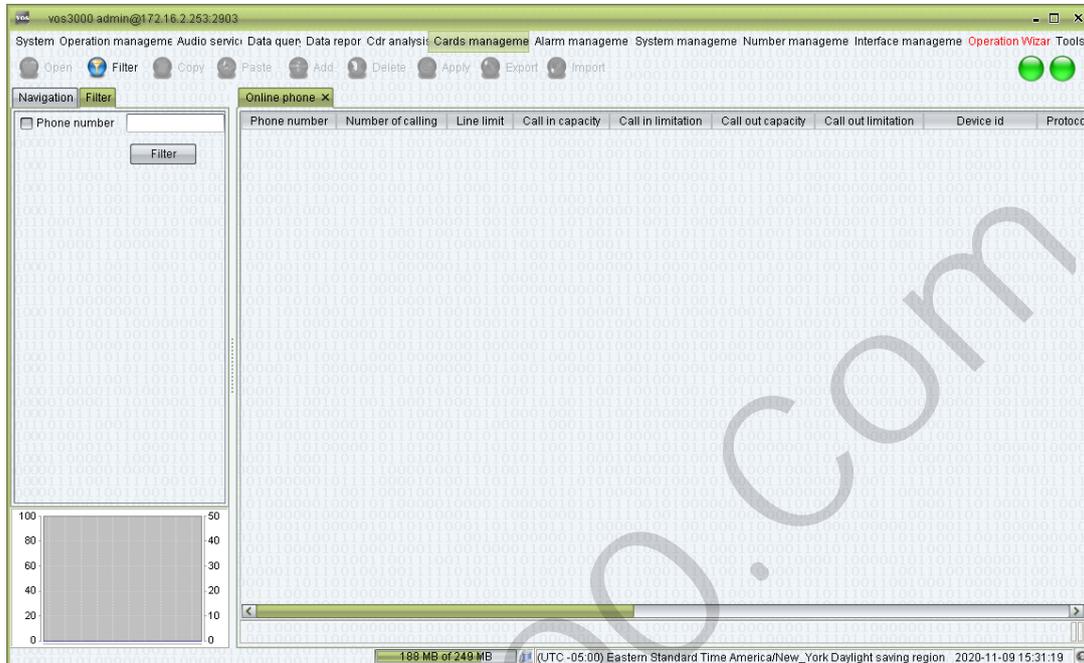
2.5.2.2 Online Phone

This function is used to query online phone.



NOTE

Static phone is not shown.



How to Start

- Double-click “Navigation > Operation management > Phone operation > Online phone”

Table Items

- Phone number: the number used by the terminal at registration (used as the caller ID and the called number).
- Number of calling
- Line limit
- Call in capacity
- Call in limitation
- Call out capacity
- Call out limitation
- Device id: Model of the device.
- Protocol: the protocol used at registration (SIP or H323).
- Registered ip: the remote address of the terminal used at the registration. If the connection is establish through a firewall, this address may be a local address.
- Registration time: the server time of the terminal’s most recent registration.
- Update time: the time of the most recent confirmation that the terminal is online.
- Duration: the time elapsed since the most recent registration.
- Encryption type: the type of encryption used by the gateway.
- Call tracing: current call tracing status.

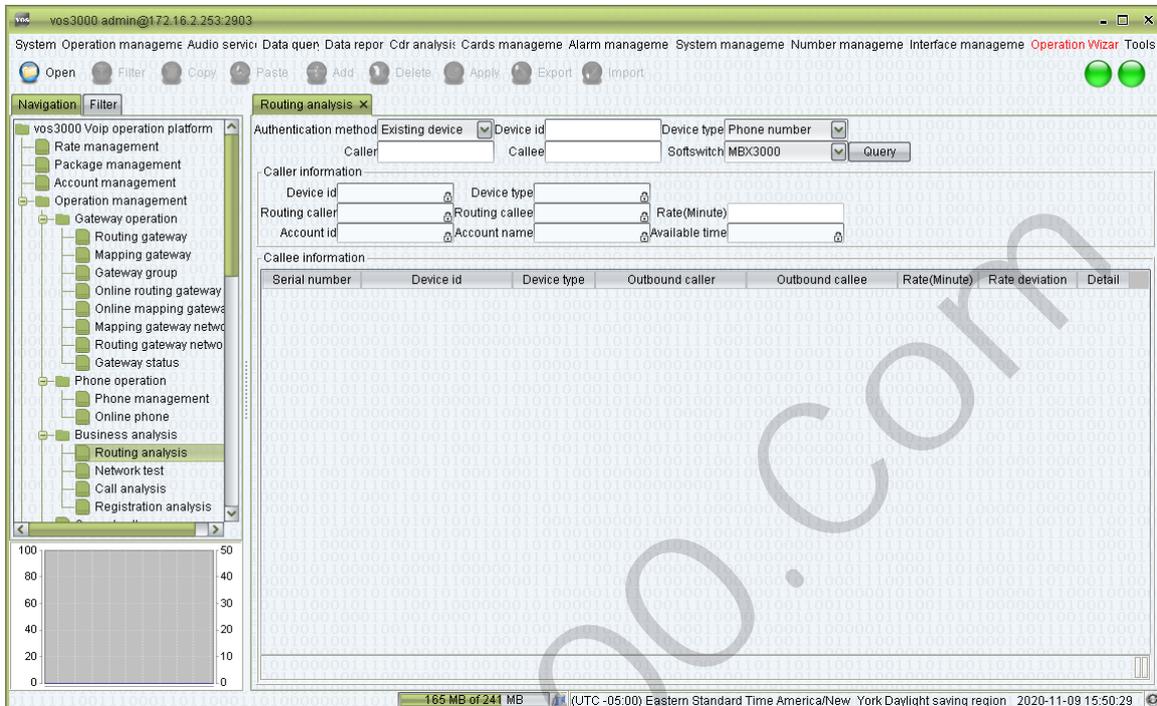
- Register tracing: current register tracing status.
- Local ip: this item is empty, call returns from the original address.
- Softswitch name: the name of the softswitch that the phone registered.

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2.5.3 Business Analysis

2.5.3.1 Routing Analysis

This function is used to simulate the routing process of phone or mapping gateway.



How to Start

- Double-click “Navigation > Operation management > Business analysis > Routing analysis”

Input Items

- Authentication method:
 - Existing device: phone or gateway.
 - Static ip: IP of mapping gateway.
- Device id: phone number or gateway ID depends on device type.
- Device type:
 - Phone number
 - Mapping gateway
- Caller: simulated caller number.
- Callee: simulated callee number.
- Softswitch: the name of the softswitch.

Output Items

Caller information

- Device id: actual caller device name.

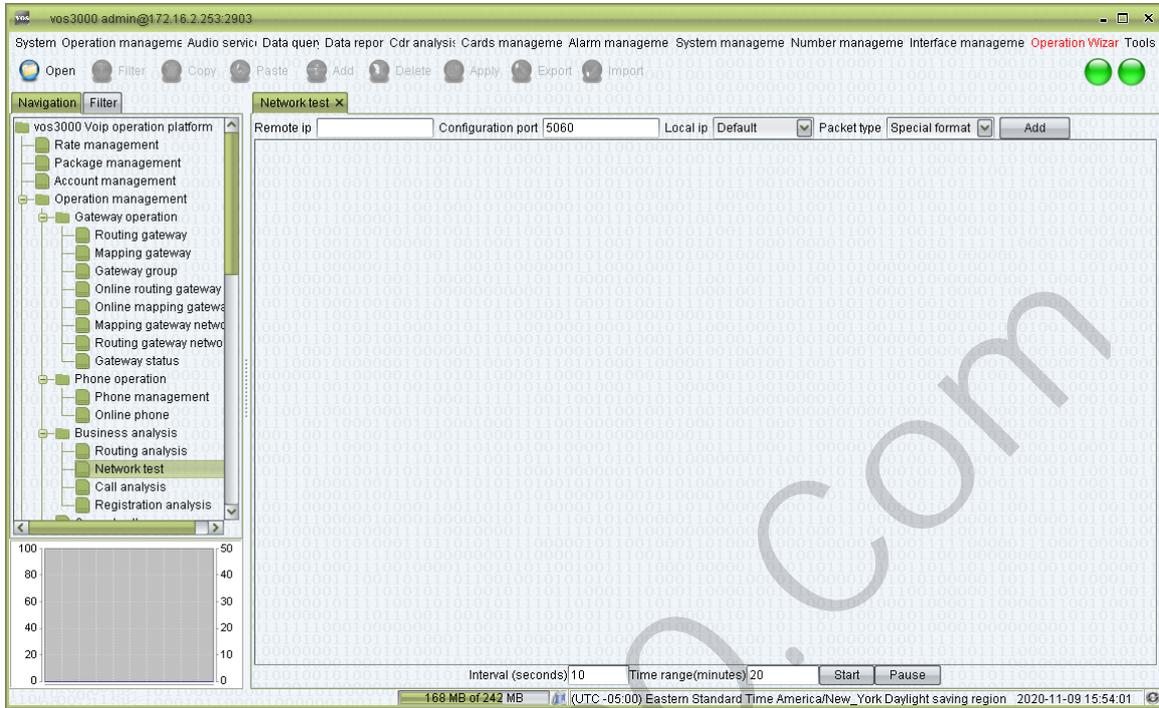
- Device type: actual caller device type.
- Routing Caller: caller after caller device's dial plan.
- Routing Callee: callee after caller device's dial plan.
- Rate(Minute): calculate by billing account's fee rate.
- Account id: billing account id.
- Account name: billing account name.
- Available time: maximum duration.

Callee information

- Serial Number: routing sequence.
- Device id: callee device name trying to call.
- Device type: callee device type trying to call.
- Outbound caller: caller after callee device's dial plan.
- Outbound callee: callee after callee device's dial plan.
- Rate(Minute): calculate by callee device's fee rate.
- Rate deviation: difference between caller device's fee rate and callee device's cost.
- Detail: like prefix, priority, score and so on.

2.5.3.2 Network Test

This function is used to test to a specified IP network condition.



How to Start

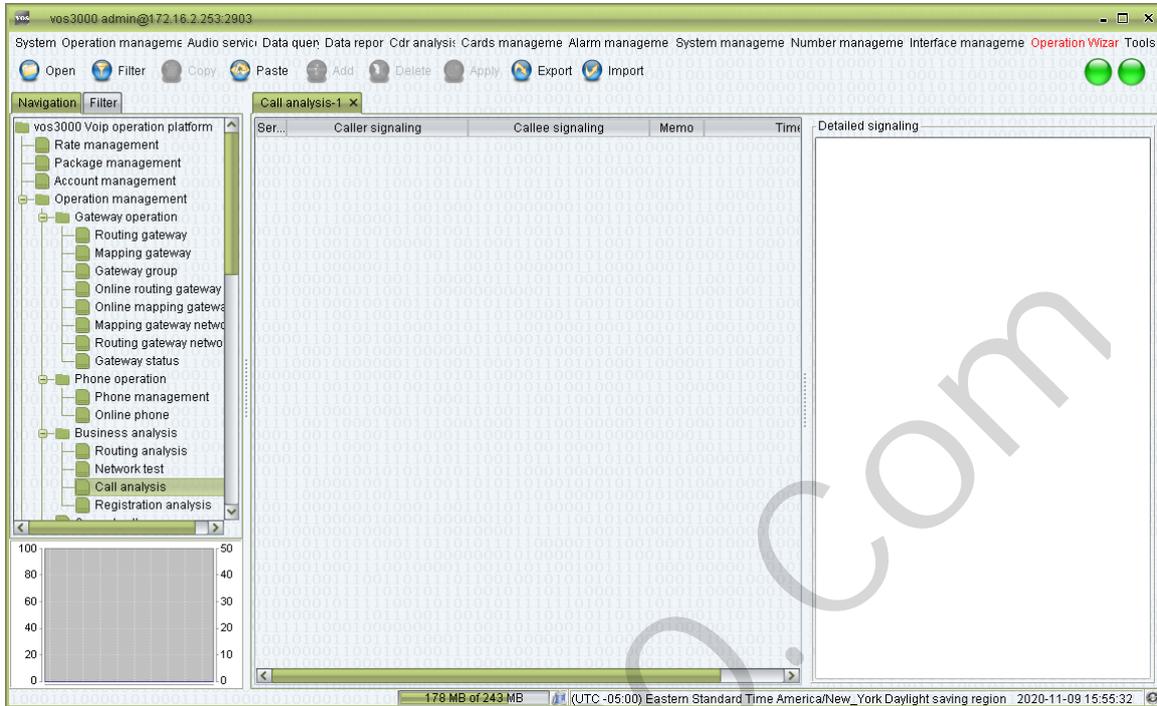
- Double-click “Navigation > Operation management > Business analysis > Network test”

Table Items

- Remote ip: ip addresses.
- Configuration port: ip port.
- Local ip: local authorized ip address.
- Packet type:
 - Special format: test VOS production.
 - ICMP: test generic network type.

2.5.3.3 Call Analysis

This function is used to analysis call problem.



How to Start

- Double-click “Navigation > Operation management > Business analysis > Call analysis”

Table Items

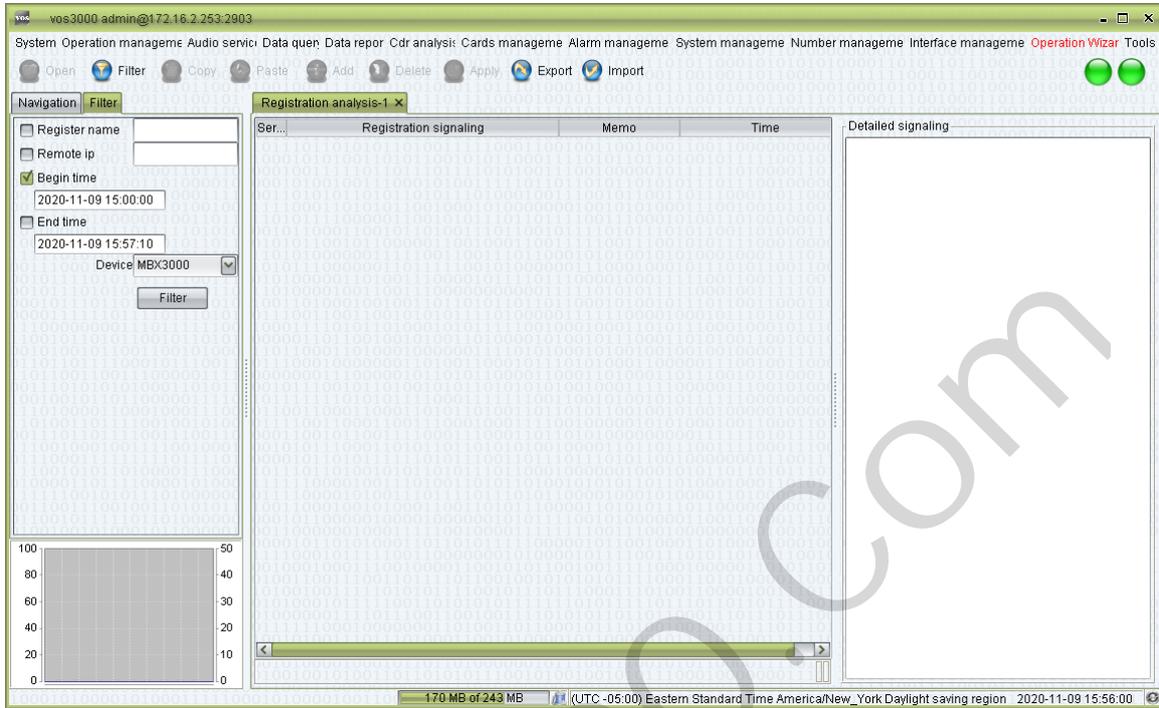
- Serial number: the serial number of signaling interaction.
- Caller signaling: content of signaling interaction with caller.
- Callee signaling: content of signaling interaction with callee.
- Memo: message of softswitch.
- Time: time of signaling.

Other Operations

- Export: save the signaling as file.
- Import: import the signaling file to do analysis.

2.5.3.4 Registration Analysis

This function is used to analysis registration problem.



How to Start

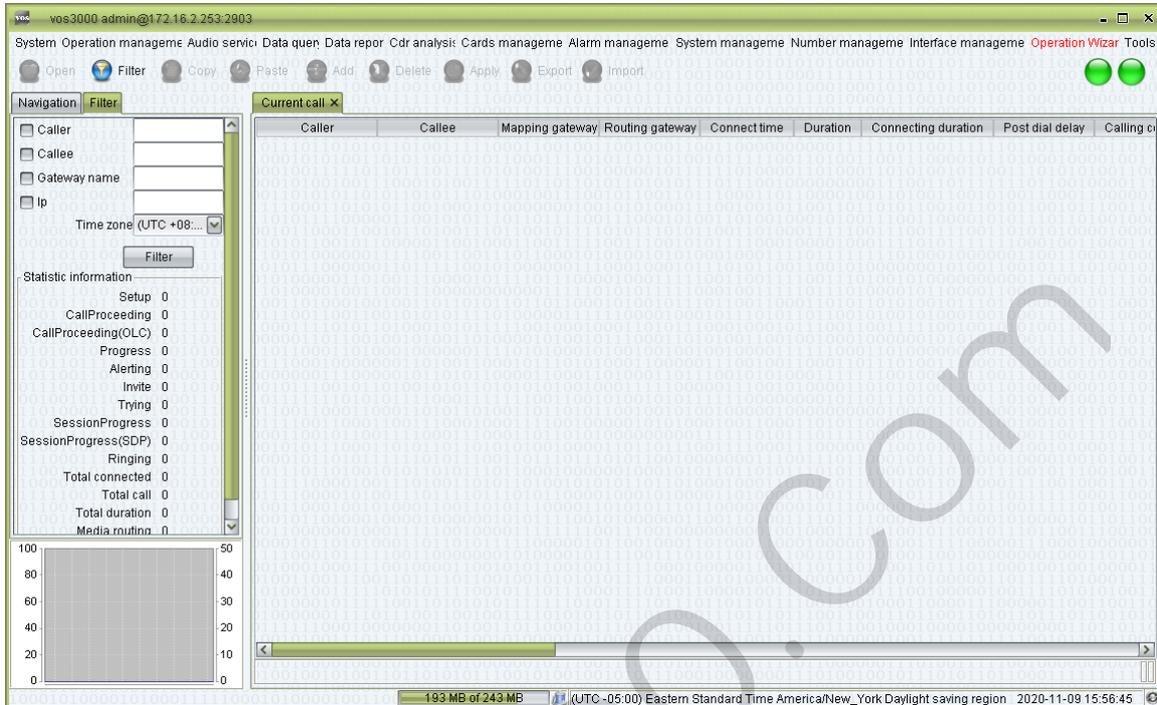
- Double-click “Navigation > Operation management > Business analysis > Registration analysis”

Table Items

- Serial number: the serial number of signaling interaction.
- Registration signaling: content of signaling interaction.
- Memo: message of softswitch.
- Time: time of signaling.

2.5.4 Current Call

This function is used to query current call.



How to Start

- Double-click “Navigation > Operation management > Current call”

Table Items

- Caller: the number of the caller.
- Callee: the number of the called.
- Mapping gateway: the gateway between the caller and the softswitch.
- Routing gateway: the gateway between the called and the softswitch.
- Connect time: the time elapsed since the establishment of the connection.
- Duration: duration of the call.
- Connecting duration: time elapsed from platform received to call connected.
- Post dial delay: time elapsed from send call to routing response.
- Calling code: the voice encoding used in the session.
- Caller rx audio: the voice traffic and package loss of caller.
- Caller rx video: the video traffic and package loss of caller.
- Callee rx audio: the voice traffic and package loss of callee.
- Callee rx video: the video traffic and package loss of callee.
- Caller coding: voice encodings supported by the caller (acquired by analyzing the signals).
- Callee coding: voice encodings supported by the called (acquired by analyzing the signals).

- Caller information: the IP and RTP IP of the caller.
- Callee information: the IP and RTP IP of the callee.
- Caller dtmf: the DTMF mode of the caller.
- Callee dtmf: the DTMF mode of the called.
- Media routing: whether the RTP is routed by servers.
- Caller device name: the manufacturer of the caller device.
- Callee device name: the manufacturer of the callee device.
- Caller encryption type: the encryption used by the caller.
- Callee encryption type: the encryption used by the called.
- Softswitch ip: the IP address of the softswitch.

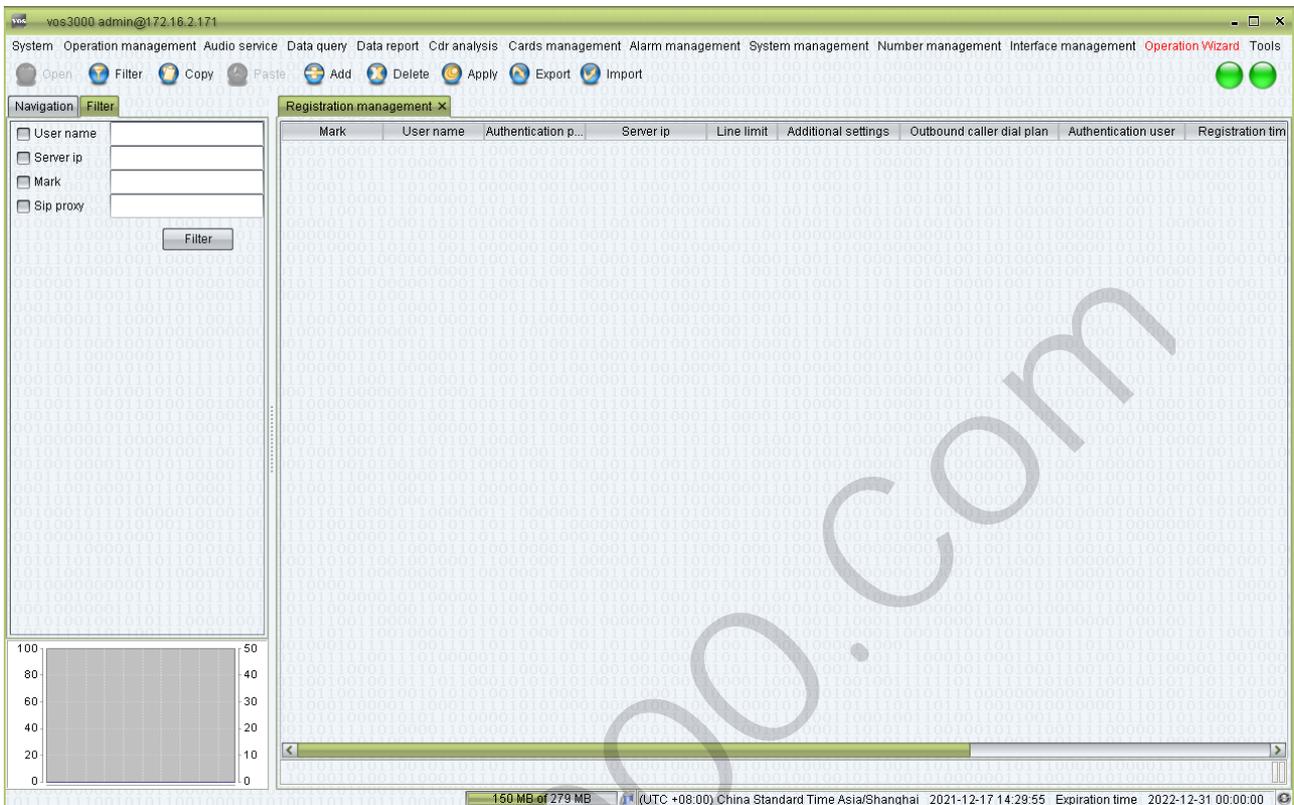
Right-Click Menu

Caller	Callee	Mapping gateway	Routing gateway	Connect time	Duration	Connecting duration	Post dial delay	Calling ci
10087	10088			05:00:42	00:00:06	1.565	0.015	Audio:PCM

- Total up calls
- Disconnect call
- Audio traffic
- Call analysis

2.5.5 Registration Management

This function is used to manage registration to other platform.



How to Start

- Double-click “Navigation > Operation management > Registration management”

Table Items

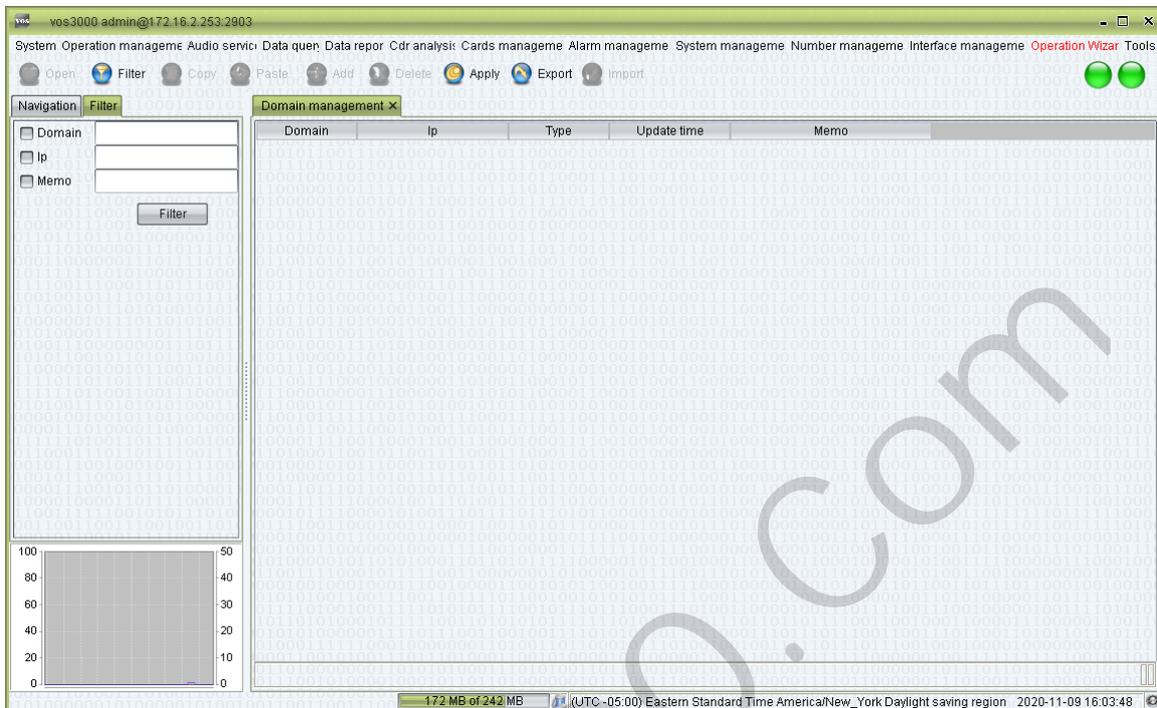
- **Mark:** identifier of registration information. When routing gateway type is registration, use mark as registration identification.
- **User name:** user name used to register to the server.
- **Authentication password:** password used to register to the server.
- **Server ip:** ip address of the registration server.
- **Line limit**
- **Additional settings:**
 - **Signaling port:** port of the registration server.
 - **Encryption:** whether to register using encryption.
 - **Host name:** server address field of SIP REGISTER/FROM/TO, use server ip as default.
 - **Sip proxy:** address of SIP Route.
 - **User-Agent:** field of SIP User-Agent, use VOS3000 and version as default.
 - **Local ip:** local ip address used to register.
 - **Local port:** local port used to register.

- Register period:
 - ◆ Default: set by “Operation management > Softswitch management > Additional settings > System parameter > SS_SIPREGISTEREXPIRE”.
 - ◆ Auto negotiation: decide by server.
- Outbound caller dial plans
- Outbound caller dial plans
- Authentication user: username field of SIP 401/407 message, use user name as default.
- Registration time: time of registration.
- Update time: time of last update.
- Actual register cycle: registration cycle after negotiation.
- Last error: error of last time.
- Softswitch

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2.5.6 Domain Management

This function is used to manage domain, which is used for Routing Gateway and Registration.



How to Start

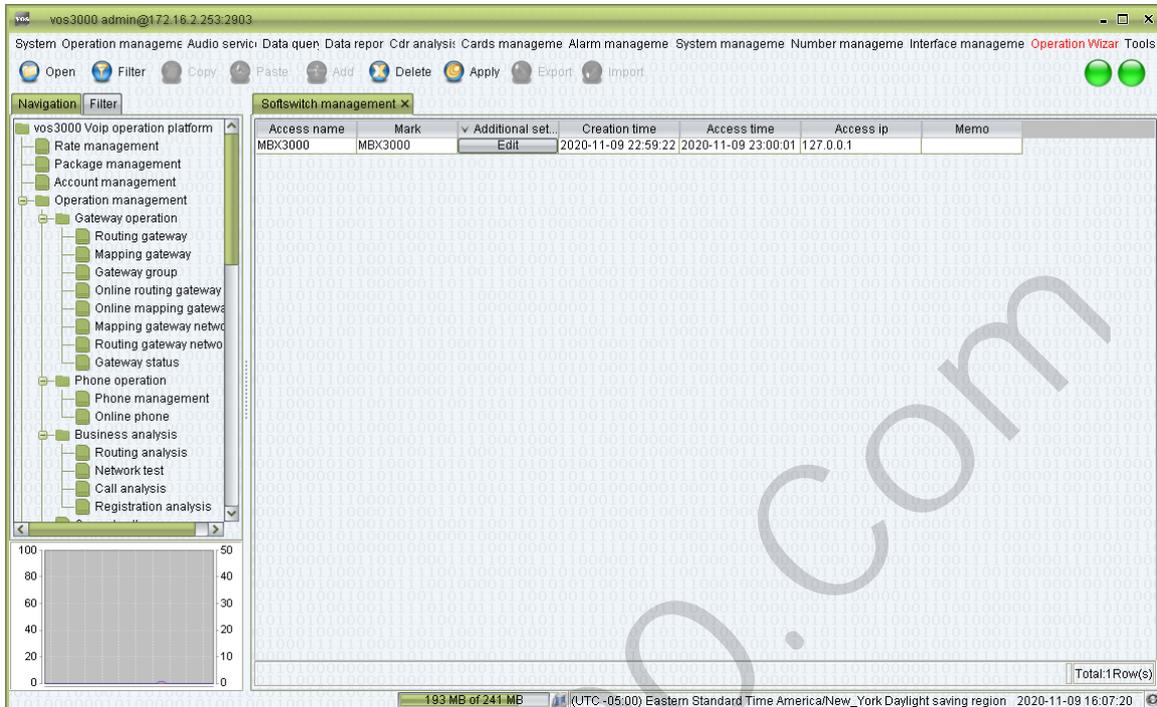
- Double-click “Navigation > Operation management > Domain management”

Table Items

- Domain: the domain name of routing gateway or registration.
- Ip: Domain’s IP.
- Type:
 - Dynamic: update DNS every several minutes, can be set by “System management > System parameter > SERVER_DNS_UPDATE_INTERVAL”.
 - Static: won’t check DNS.
- Update time: DNS last update time.
- Memo

2.5.8 Softswitch Management

This function is used to manage softswitch.



How to Start

- Double-click “Navigation > Operation management > Softswitch management”

Table Items

- Access name: the name of the Softswitch.
- Mark: named by the management platform.
- Additional settings
- Creation time: the time of first access to the softswitch.
- Access time: the most recent access to the softswitch.
- Access ip: the IP address of the softswitch.
- Memo: comments on the softswitch.

Right-Click Menu

- Synchronize data: synchronize settings of the softswitch with VOS3000.
- Current call: current sessions on the softswitch.
- System information: information about the softswitch.

The screenshot displays the VOS3000 administration interface. The main window is titled 'vos3000 admin@172.16.2.253:2903'. The navigation pane on the left shows a tree structure with categories like 'Rate management', 'Package management', 'Account management', 'Operation management', 'Gateway operation', 'Phone operation', and 'Business analysis'. The 'Operation management' section is expanded to show 'Softswitch management', which is further expanded to 'Softswitch+MBX3000+system information'. The main content area displays a table of system parameters and their values.

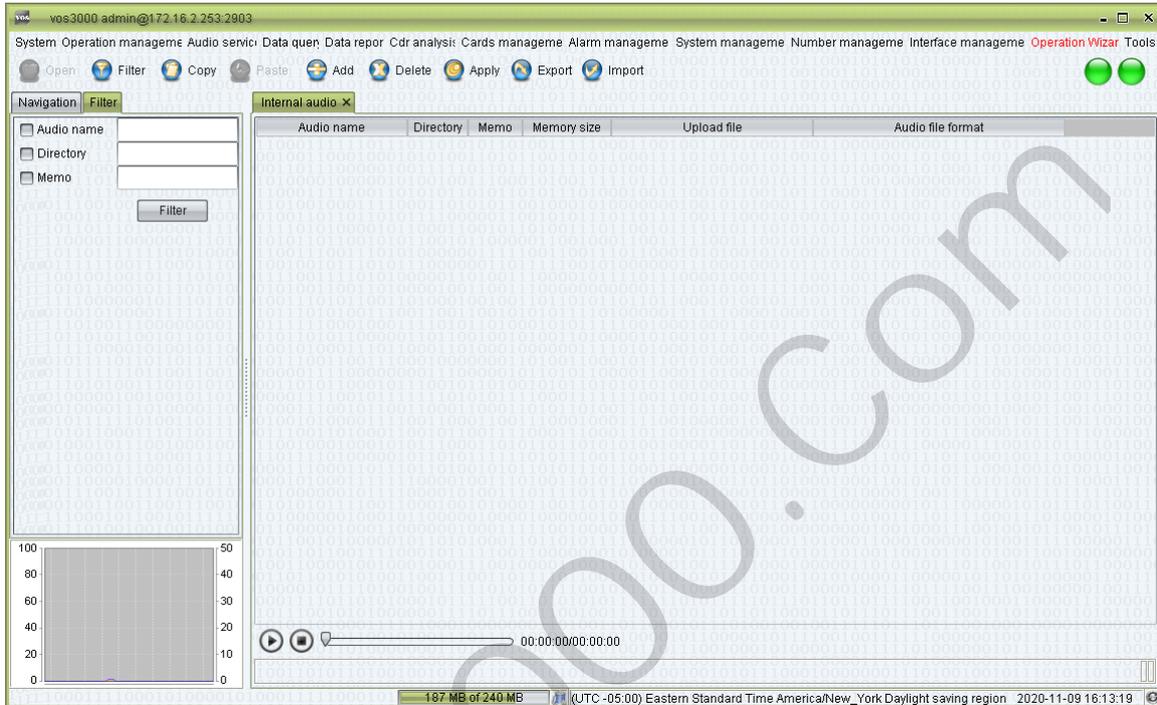
Information name	Information value
License	
PROTOCOL	H323/SIP
VALID_DATE	PERMANENT
TERMINAL_LIMIT	NO_LIMIT
CALL_LIMIT	5000
System Parameter	
VERSION	V2.1.9.02 Build Sep 29 2020 14:50:22
START_TIME	Nov 9 2020 09:59:57
AUTHORIZED_IP	172.16.2.253
BIND_IP	172.16.2.253
SIP_PORT	5060,6060
SIP_RC4_PORT	5070
SIP_STATIC_RC4_PORT	5078
SIP_STATIC_AES128_PORT	5079
SIP_STATIC_AES256_PORT	5080
SIP_DYNAMIC_XOR_RC4_PORT	5071
SIP_DYNAMIC_XOR_AES128_PORT	5072
SIP_DYNAMIC_XOR_AES256_PORT	5073
SIP_DYNAMIC_RC4_PORT	5074
SIP_DYNAMIC_AES128_PORT	5076
SIP_DYNAMIC_AES256_PORT	5077
Audio Player	
AUDIO_PLAYER_ONLINE	
Trace Status	
TRACE_CALL	OFF
TRACE_REGISTER	OFF

At the bottom of the interface, there is a status bar showing '177 MB of 241 MB' and the time '(UTC -05:00) Eastern Standard Time America/New_York Daylight saving region 2020-11-09 16:10:34'. A watermark 'VOS3000.COM' is visible across the screenshot.

2.6 Audio Service

2.6.1 Internal Audio

This function is used to manage error prompt audio file.



How to Start

- Double-click “Navigation > Audio service > Internal audio”

Table Items

- Audio name: custom name of the voice file added by the user.



NOTE

Special characters like “\|*?<>|!#%^&” and “..” are not allowed for the name.

- Directory: the language catalogue of the audio.
- Memo: comments on the voice file.
- Memory size: display of the size of the voice file. (“0” indicates the file has not been uploaded yet.)
- Upload file: specify the local path of the voice file for upload.
- Audio file format: display the format of the uploaded voice file.

Toolbar

- Filter: filter the items.
- Copy: copy the selected audio information.



NOTE

The actual audio file data will not be copied.

- Add: add contents to the service.
 - Delete: delete selected contents.
 - Import: import voice files in batch mode. The voices will be automatically named according to their file names.
-



CAUTION

Supported formats include WAV files (8 KHz, 16 bit, monophonic).

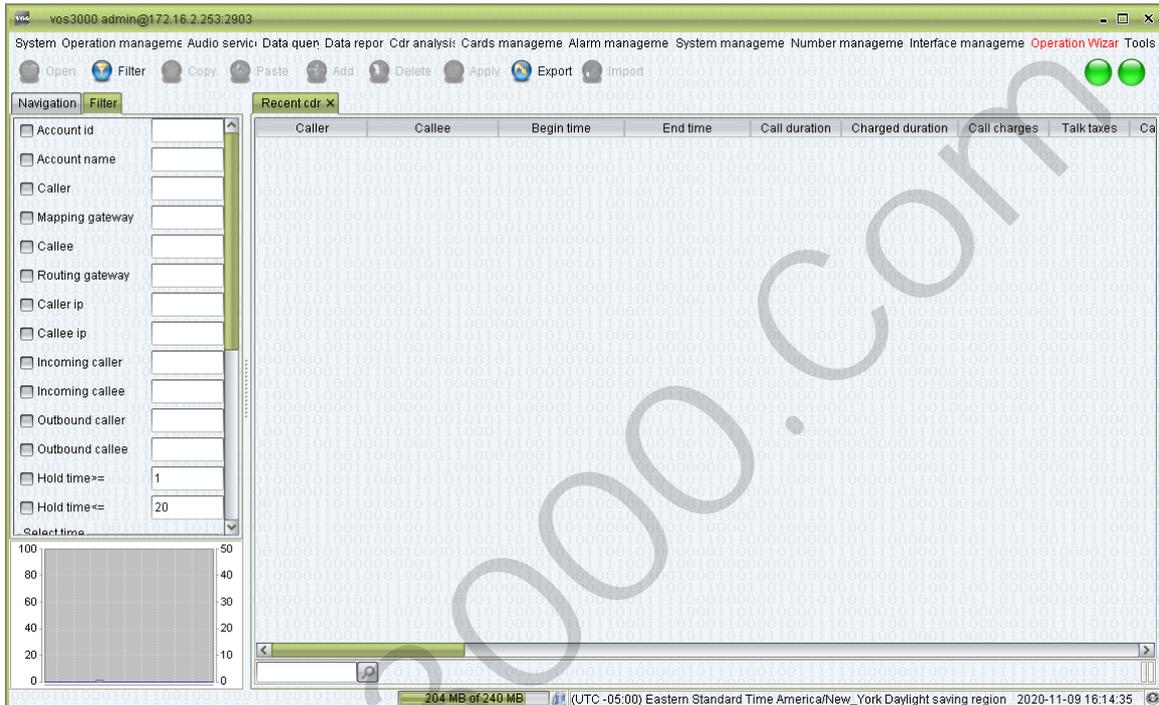
VOS3000.Com

2.7 Data Query

Historical data is independent, won't be affected by configurations, e.g. delete accounts won't remove account's CDR.

2.7.1 Recent CDR

This function is used to query recent CDR.



How to Start

- Double-click “Navigation > Data query > Recent Cdr”

Table Items

See “CDR”.

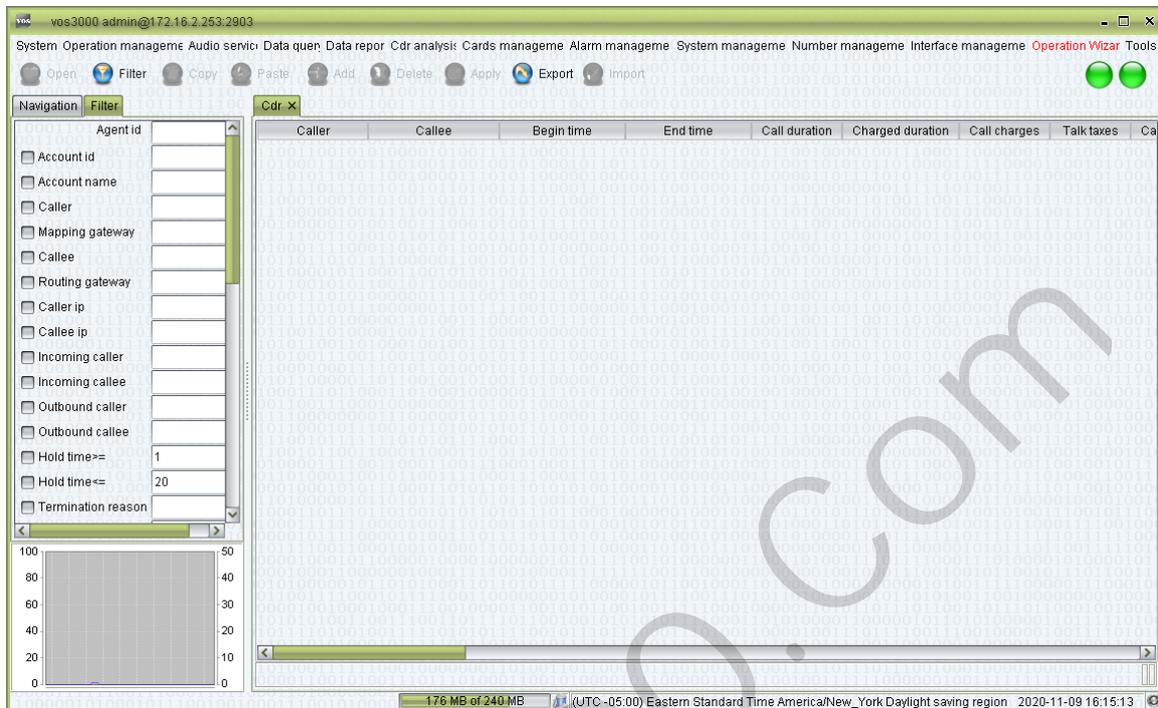


NOTE

This function is used to quickly query out the last 1000 CDR.

2.7.2 CDR

This function is used to query CDR.



How to Start

- Double-click “Navigation > Data query > Cdr”

Table Items

By default, the system displays 1000 records every page. This number can be changed in “System parameter > SERVER_QUERY_ONE_PAGE_SIZE”.

- Caller: the phone number of the caller.
- Callee: the phone number of the caller.
- Begin time: the time when the call is initiated.
- End time: the time the call lasts.

NOTE

In Normal Mode, End Time is begin time + conversation time.

In Clearing Mode, End Time is the actual stop time.

Accuracy of billing time makes the difference, e.g. call from 201304018:13:34:40.002 to 201304018:13:34:41.003, billing duration 1.001 second, according to accuracy should treat as 2 second, and display 201304018:13:34:42.002.

- Call duration: Actual duration.
- Charged duration: the time used for billing, which is calculated according the Billing Cycle specified in rate policies.
- Call charges: the fee charged for this call.
- Talk taxes: the charge of taxes for this call.
- Call expense: the cost of delivering this call.

- Cost taxes: the cost of taxes for this call.
- Termination reason: see Appendix for details.
- Hangup side: call termination initiator.
- Mapping gateway: the ID of the gateway between the caller and the Softswitch.
- Routing gateway: the ID of the gateway between the called and the Softswitch.
- Caller ip: the IP address of the caller.
- Callee ip: the IP address of the called.
- Account name: the name of the account used for billing this call.
- Account id: the number of the account used for billing this call.
- Agent id: the agent number corresponding to the billing account number used in this call.
- Call type: “Network”, “Local”, “Domestic” or “International”.



NOTE

Call type depends on the matched fee rate’s Rate Type.

- Area prefix: the prefix used for billing this call.
- Area name: obtain the name of the prefix according to the area prefix.
- Incoming caller: original caller number sent to server.
- Incoming callee: original callee number sent to server.
- Outbound caller: the caller number sent to the called after the application of dial plans.
- Outbound callee: the called number sent to the called after the application of dial plans.
- Caller device name
- Callee device name
- Package duration: duration of the package used.
- Package charges: fees of the packages used.
- Billing method:
 - By caller: billing on caller account.
 - By callee: billing on callee account.
- Billing mode: billing caller device type including “Phone, Gateway, Phone card”.
- Continue duration: Time elapsed from platform received to call connected.
- Connect delay: Time elapsed from send call to routing response.
- Calling Call-Id: call identification within the caller signaling.
- Called Call-Id: call identification within the callee signaling.
- Reason: Vos will add “reason” filed in hangup signaling, used to transparent peer’s ”reason” filed which includes hangup reason.
- Serial number: unique identification.



NOTE

PDD is defined as the time from send call to callee to receive signal (switch routing gateway will recalculate), see detail below:

Callee is SIP

1. Receive callee’s 180/200
2. Receive callee’s 486/600
3. If callee is phone, callee’s signal contains SDP

4. If callee is routing gateway and enabled “Stop Switch Gateway After Receive SDP”, callee’s signal contains SDP

Callee is H323

1. Receive callee’s Progress/Alerting/Connect
2. Receive callee’s ReleaseComplete (UserBusy)
3. Callee is phone, callee’s Q931 contains faststart or H245 contains openlogicalchannel
4. Callee is routing gateway and enabled “Stop Switch Gateway After OLC”, callee’s Q931 contains faststart or H245 contains openlogicalchannel

Media Proxy Enabled

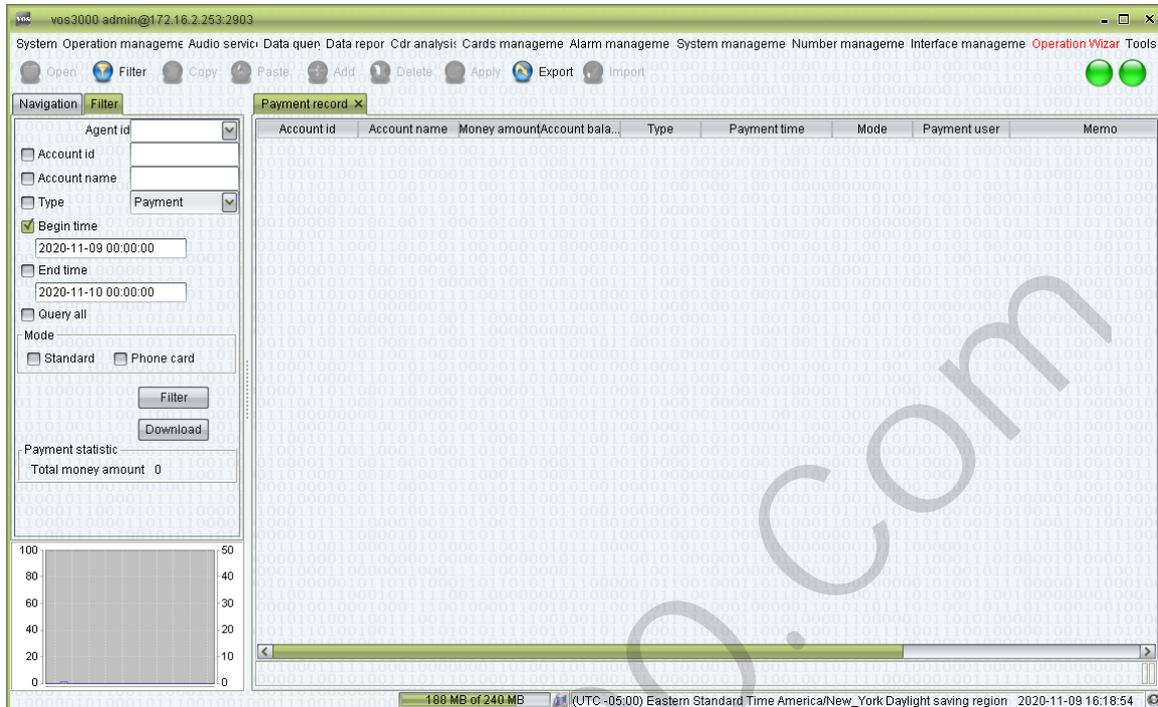
1. Callee is phone, receive callee’s RTP
2. Callee is routing gateway and enabled “Stop Switch Gateway when RTP Start”, receive callee’s RTP

Right-Click Menu

- Call analysis: open the “Call analysis” page.
- Routing gateway list: Log failed routing gateways

2.7.3 Payment Record

This function is used to query payment.



How to Start

- Double-click “Navigation > Data query > Payment record”

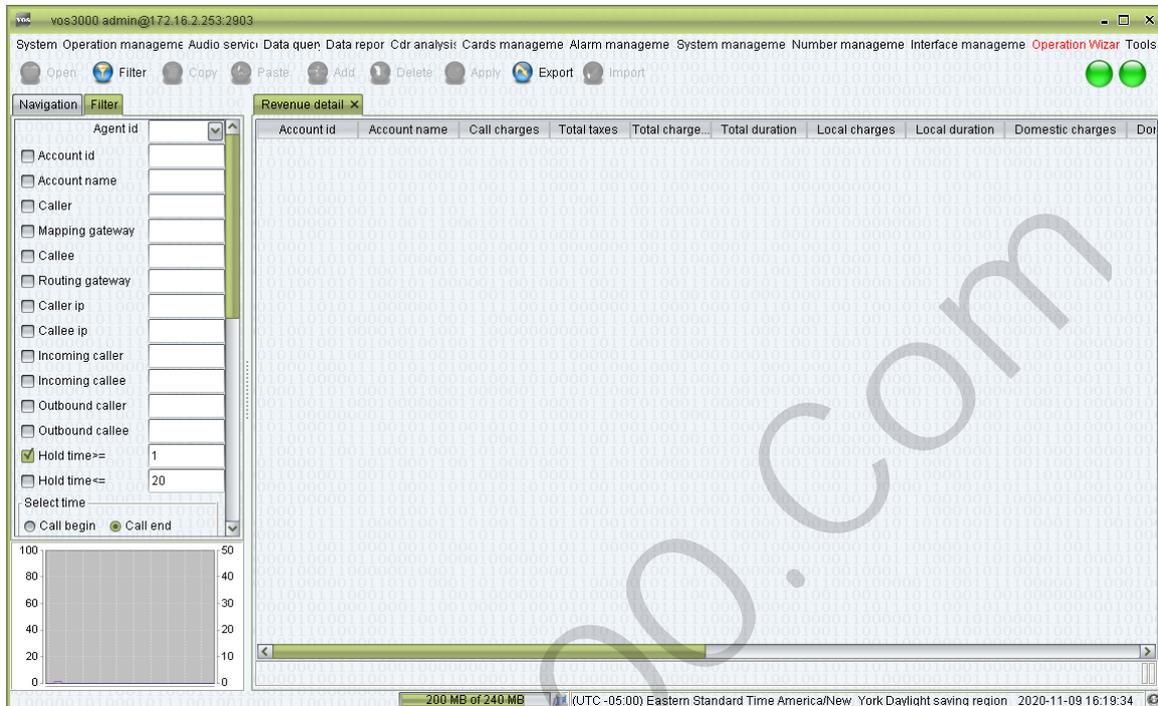
Table Items

- Account id: the paid account id.
- Account name: the paid account name.
- Payment amount: the amount.
- Account balance: the balance after payment.
- Type: “Create Account”, “Credit” or “Payment”.
- Payment time: time of the payment.
- Mode: method of the payment.
- Payment user: the name of the user that fulfills this payment.
- Memo: comments on the payment.
- Agent id: agent id for the paid account belongs.
- Agent name: agent name for the paid account belongs.
- Serial number: unique identification.

2.7.4 Bill Query

2.7.4.1 Revenue Details

This function is used to query account's consumption.



How to Start

- Double-click “Navigation > Data query > Bill query > Revenue details”

Table Items

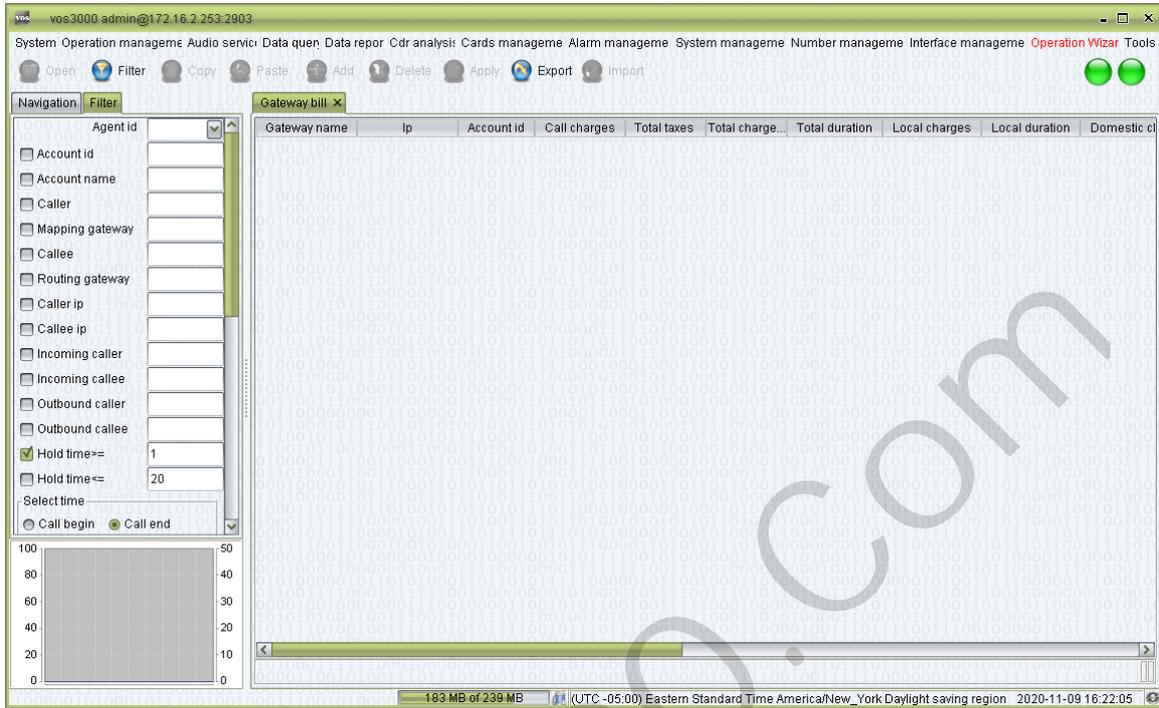
- Account id
- Account name
- Call charges
- Total taxes
- Total charges & taxes
- Total duration
- Local charges: total charge of CDRs which call type is Local.
- Local duration: total duration of CDRs which call type is Local.
- Domestic charges: total charge of CDRs which call type is Domestic.
- Domestic duration: total duration of CDRs which call type is Domestic.
- International charges: total charge of CDRs which call type is International.
- International duration: total duration of CDRs which call type is International.
- Intranet charges: total charge of CDRs which call type is Net.
- Intranet duration: total duration of CDRs which call type is Net.
- Total package: total charge of package's free money amount.

- Package duration: total duration of package's free duration.
- Number of cdr: total number of CDRs that have call length during the statistical cycle.

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2.7.4.2 Gateway Bill

This function is used to query mapping gateway's consumption.



How to Start

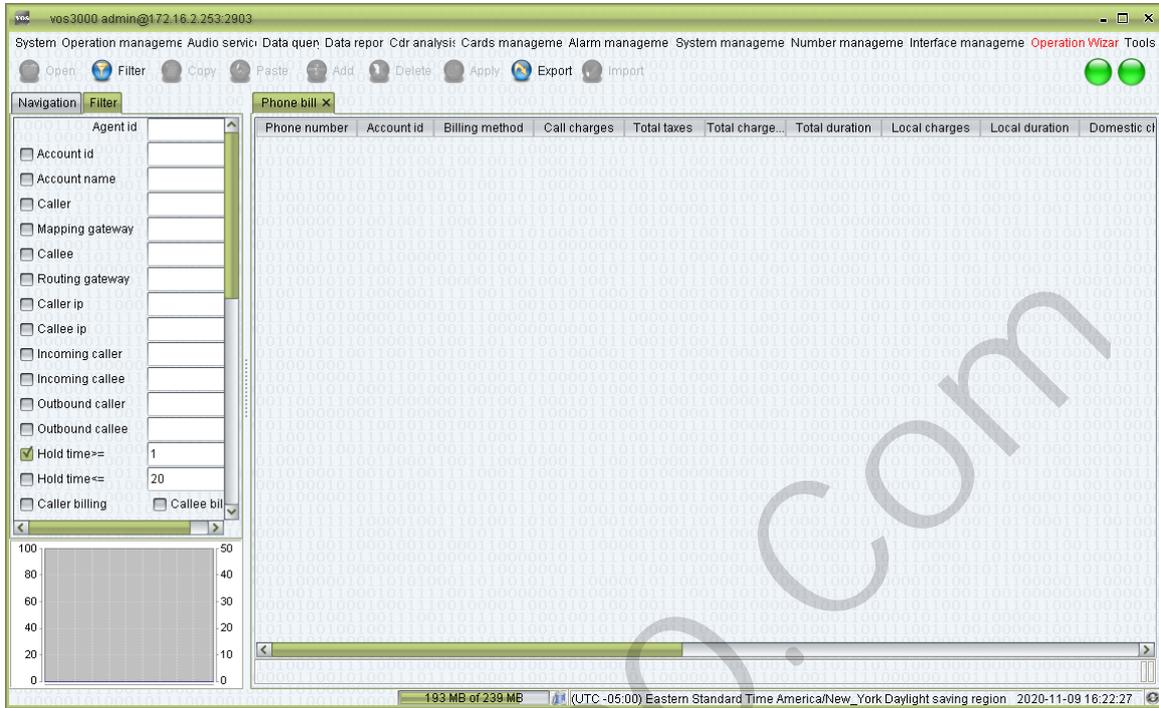
- Double-click “Navigation > Data query > Bill query > Gateway bill”

Table Items

- Gateway name: corresponding to “Gateway name” in “Mapping gateway”.
- Ip: address of caller.
- Account id: account number to which the gateway name belongs.

2.7.4.3 Phone Bill

This function is used to query phone's consumption.



How to Start

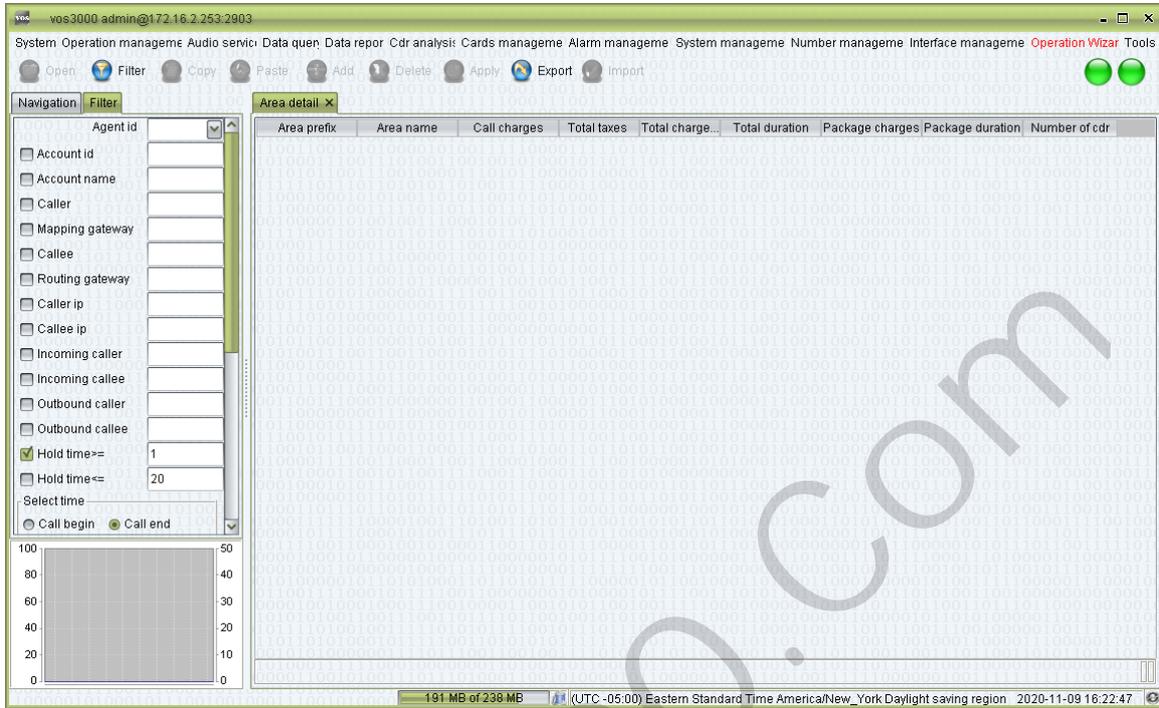
- Double-click “Navigation > Data query > Bill query > Phone bill”

Table Items

- Phone number: corresponding to “Phone number” in “Phone management”.
- Account id
- Billing method

2.7.4.4 Area Details

This function is used to query details of each area.



How to Start

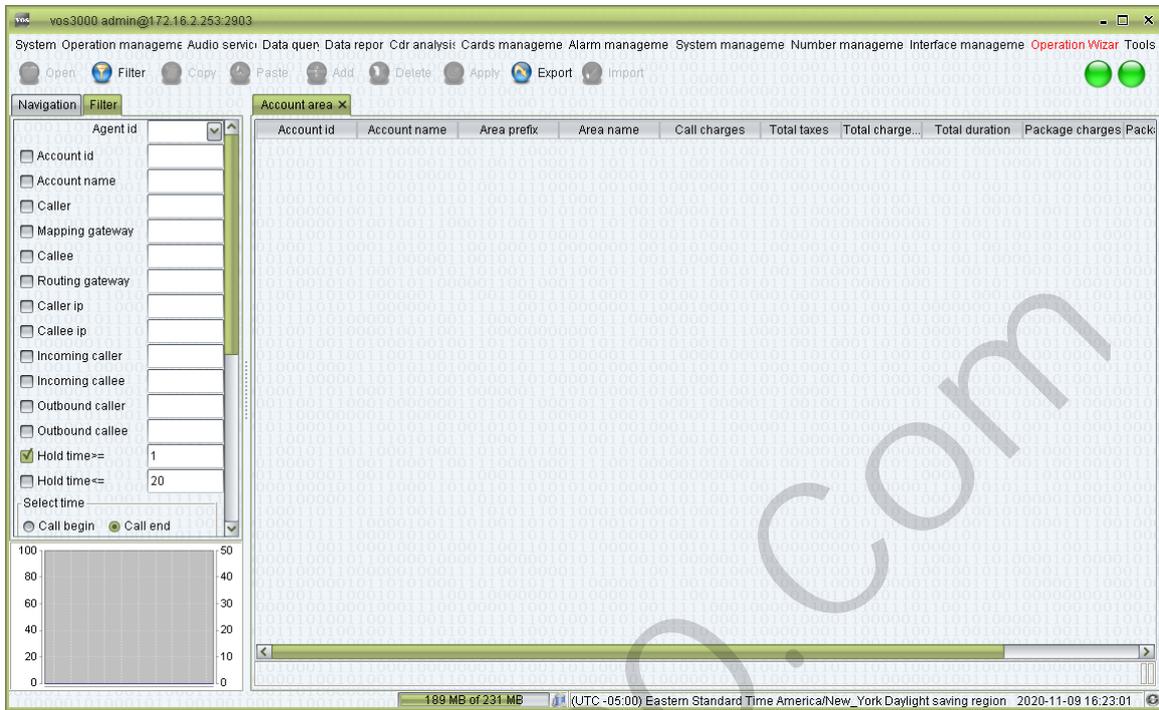
- Double-click “Navigation > Data query > Bill query > Area details”

Table Items

- Area prefix
- Area name

2.7.4.5 Account Area

This function is used to query area consumption of account.



How to Start

- Double-click “Navigation > Data query > Bill query > Account area”

Table Items

- Area prefix
- Area name

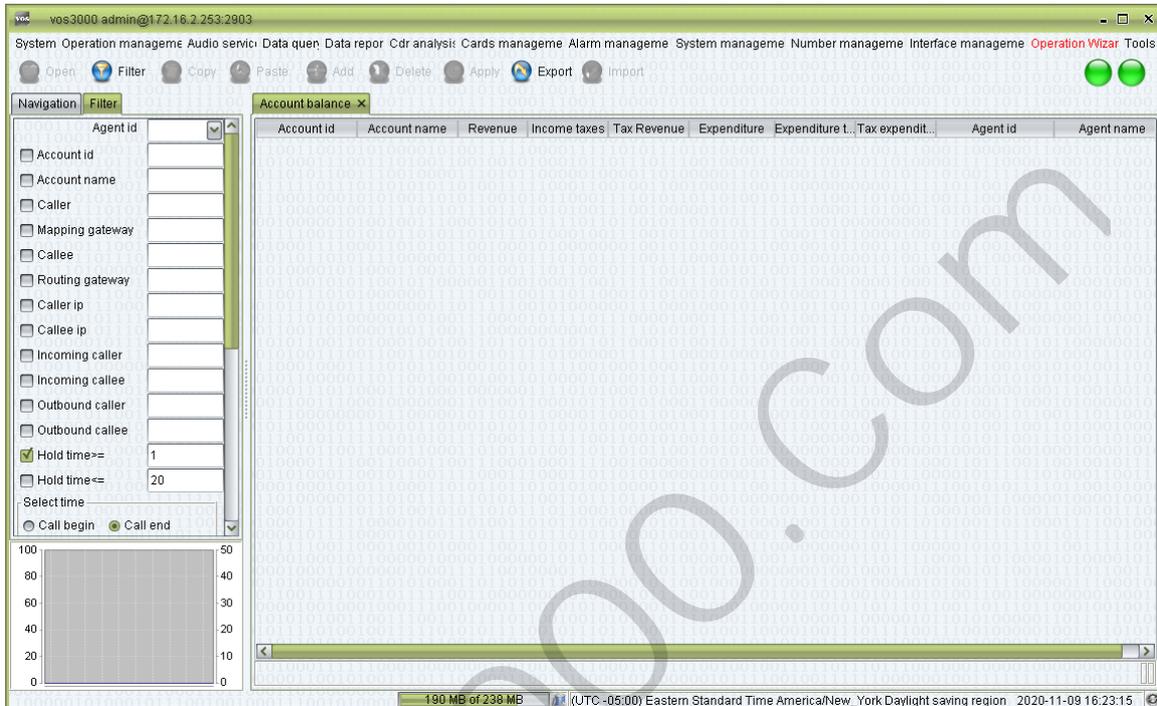
2.7.4.6 Account Balance

This function is used to query account and agent’s revenue and expenditure.

 **NOTE**

Revenue is only for agent, which is the total expenditure of sub accounts.

For ordinary accounts, there will only be expenditures in the table.



How to Start

- Double-click “Navigation > Data query > Bill query > Account balance”

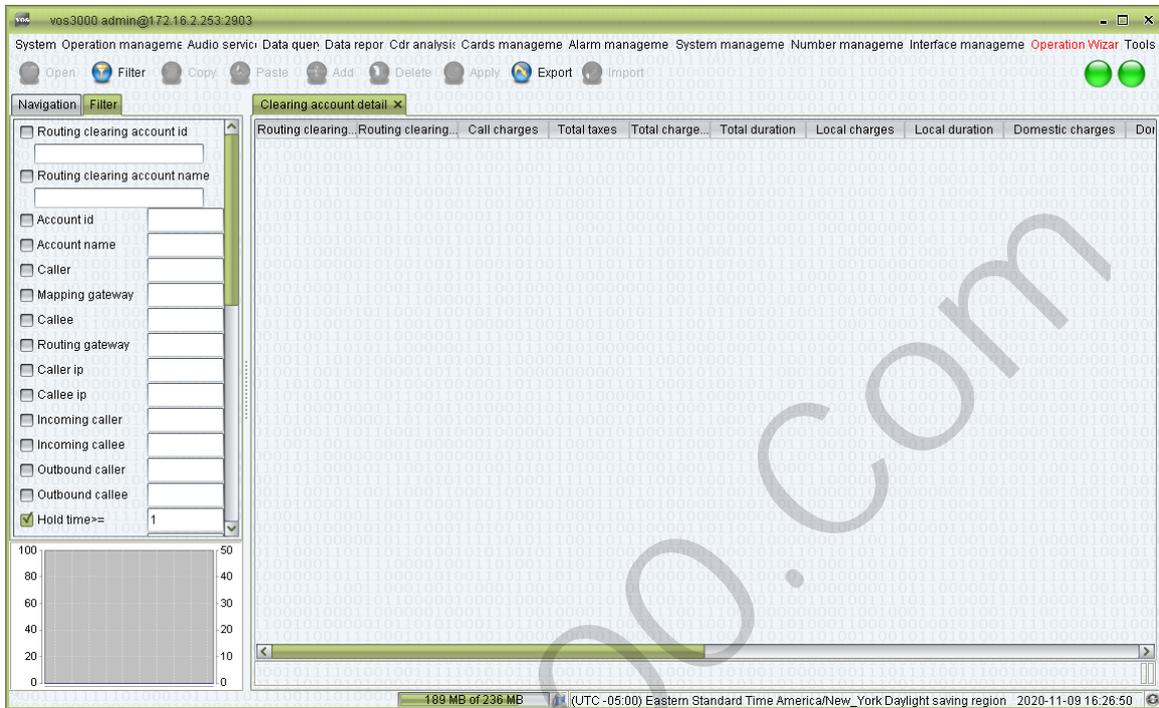
Table Items

- Revenue
- Income taxes
- Taxes revenue
- Expenditure
- Expenditure taxes
- Tax expenditures
- Agent id
- Agent name

2.7.5 Clearing Query

2.7.5.7 Clearing Account Detail

This function is used to query clearing account's consumption.



How to Start

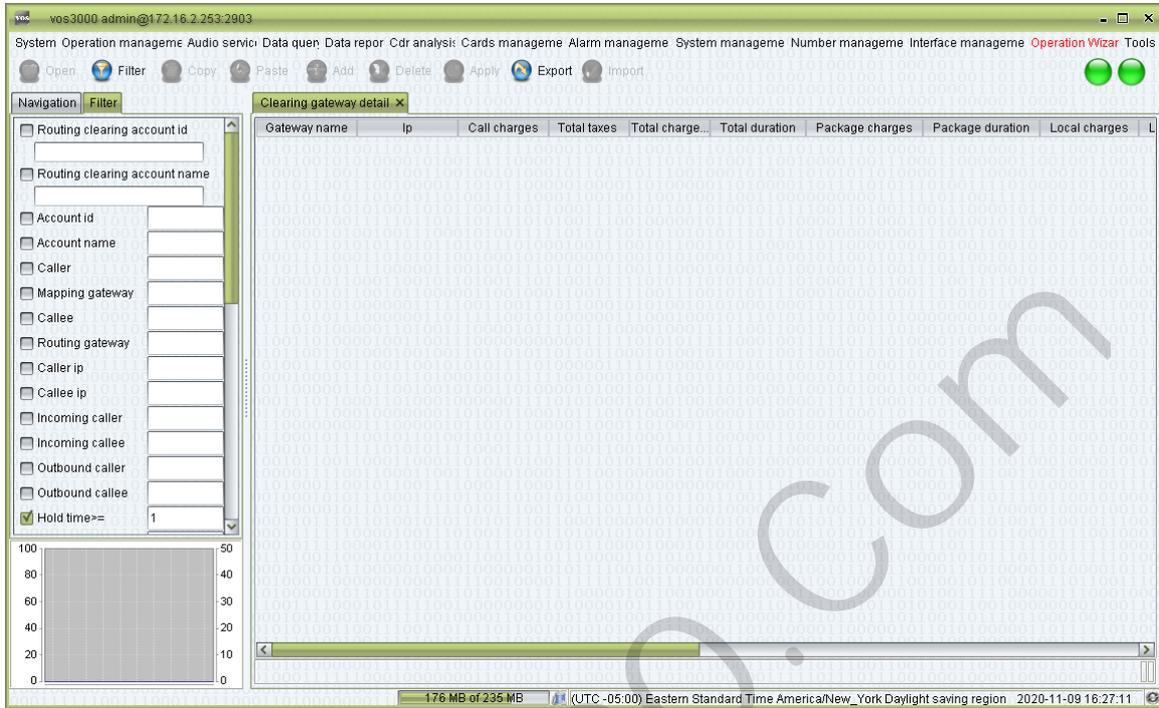
- Double-click “Navigation > Data query > Clearing query > Clearing account detail”

Table Items

- Routing clearing account id
- Routing clearing account name

2.7.5.8 Clearing Gateway Details

This function is used to query routing gateway's consumption.



How to Start

- Double-click “Navigation > Data query > Clearing query > Clearing gateway detail”

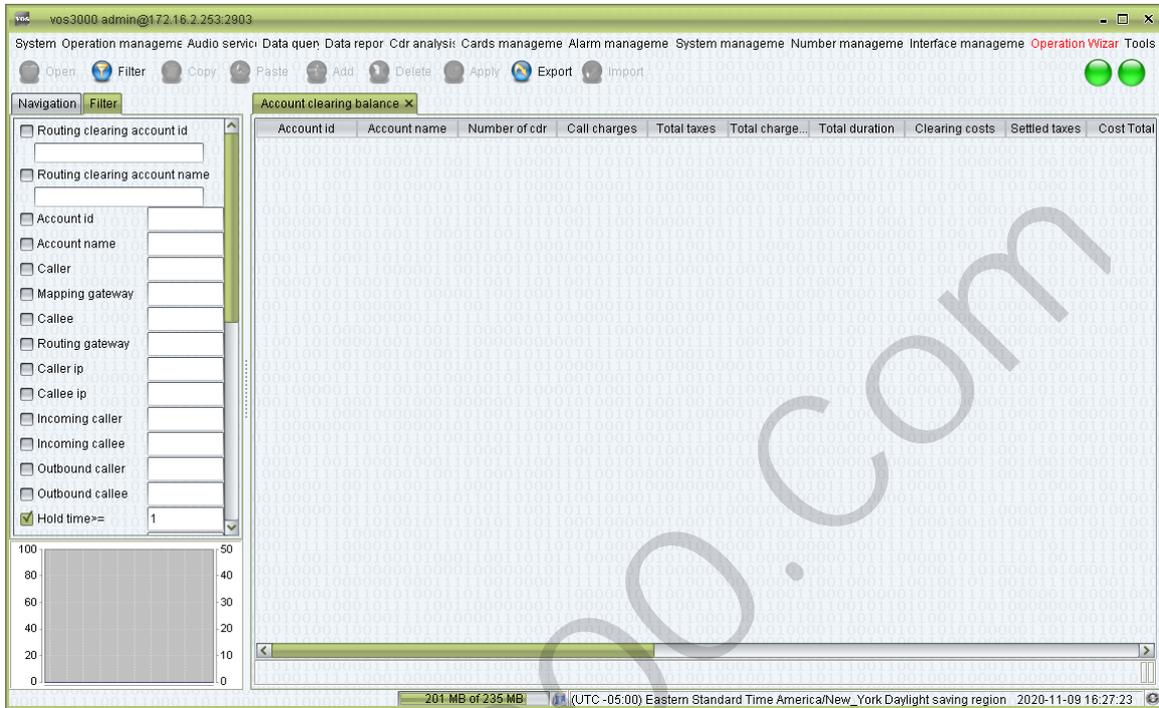
Table Items

- Gateway name
- Ip

2.7.5.1 Account Clearing Balance

This function is used to query account’s clearing consumption.

Display the expenditures on each account on different clearing account in order to quickly calculate operating profits.



How to Start

- Double-click “Navigation > Data query > Clearing query > Account clearing balance”

Table Items

- Clearing costs: total charge of clearing account.
- Clearing duration: total duration of clearing account.
- Clearing package amount: total package amount of clearing account.
- Clearing package duration: total package duration of clearing account.

2.8 Data Report

Users can specify in the “System parameter” whether to generate certain data report.

SERVER_REPORT_AGENT_INCOME	Off	Automatically generates agent income report
SERVER_REPORT_CLEARING_CUSTOMER_FEE	Off	Automatically generate clearing account detail report
SERVER_REPORT_CLEARING_CUSTOMER_IO	Off	Automatically generate account clearing balance report
SERVER_REPORT_CLEARING_CUSTOMER_LOCATION_FEE	On	Automatically generate clearing account area report
SERVER_REPORT_CLEARING_GATEWAY_FEE	Off	Automatically generate clearing gateway detail report
SERVER_REPORT_CUSTOMER_FEE	On	Automatically generate revenue detail report
SERVER_REPORT_CUSTOMER_IO	Off	Automatically generate account balance report
SERVER_REPORT_CUSTOMER_LOCATION_FEE	On	Automatically generate account area report
SERVER_REPORT_GATEWAY_CROSS_LOCATION_ASR_ACD	Off	Automatically generate gateway cross area analysis report
SERVER_REPORT_GATEWAY_FEE	On	Automatically generate gateway bill report
SERVER_REPORT_GATEWAY_MAPPING_ASR_ACD	Off	Automatically generate mapping gateway analysis report
SERVER_REPORT_GATEWAY_MAPPING_LOCATION_ASR_ACD	Off	Automatically generate mapping gateway area analysis report
SERVER_REPORT_GATEWAY_ROUTING_ASR_ACD	Off	Automatically generate routing gateway analysis report
SERVER_REPORT_GATEWAY_ROUTING_LOCATION_ASR_ACD	Off	Automatically generate routing gateway area analysis report
SERVER_REPORT_PHONE_CARD_E164_FEE	Off	Automatically generate bind number bill report
SERVER_REPORT_PHONE_CARD_FEE	Off	Automatically generate phone card bill report
SERVER_REPORT_PHONE_FEE	On	Automatically generate phone bill report



NOTE

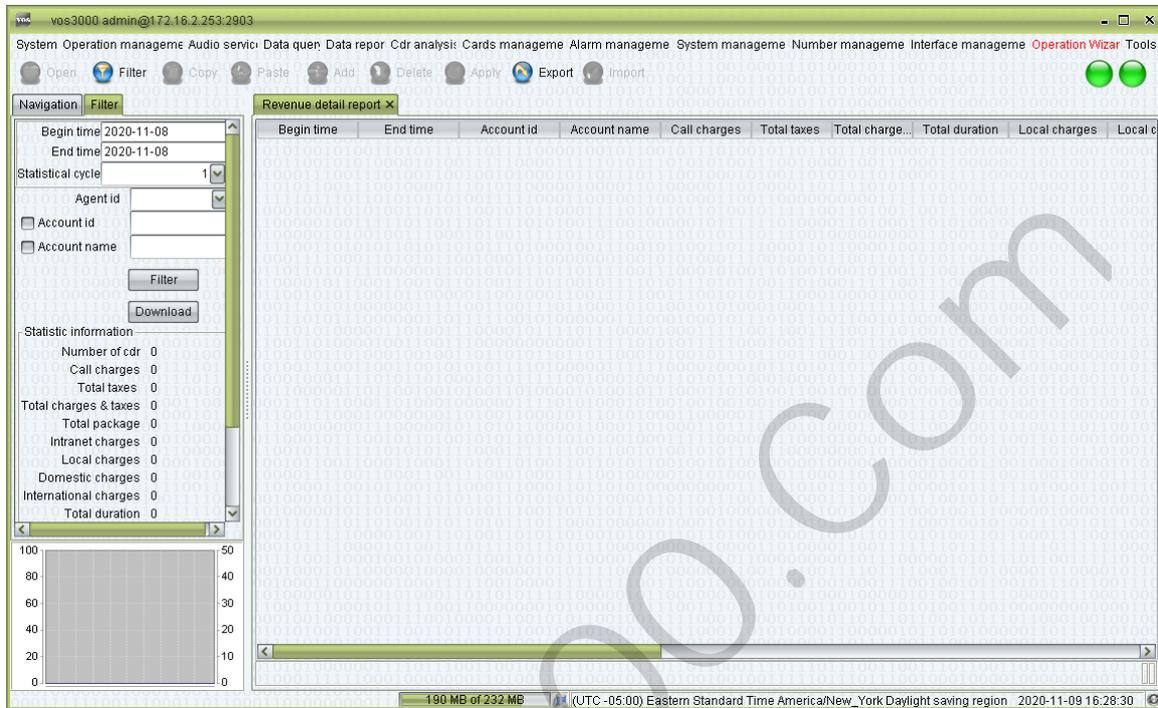
The generation of reports will begin at 1:00 A.M every day.

The time of completion depends on the capacity of the server and the amount of data.

2.8.1 Bill Report

2.8.1.1 Revenue Details Report

This function is used to query account's consumption report.



How to Start

- Double-click “Navigation > Data report > Bill report > Revenue detail report”

Table Items

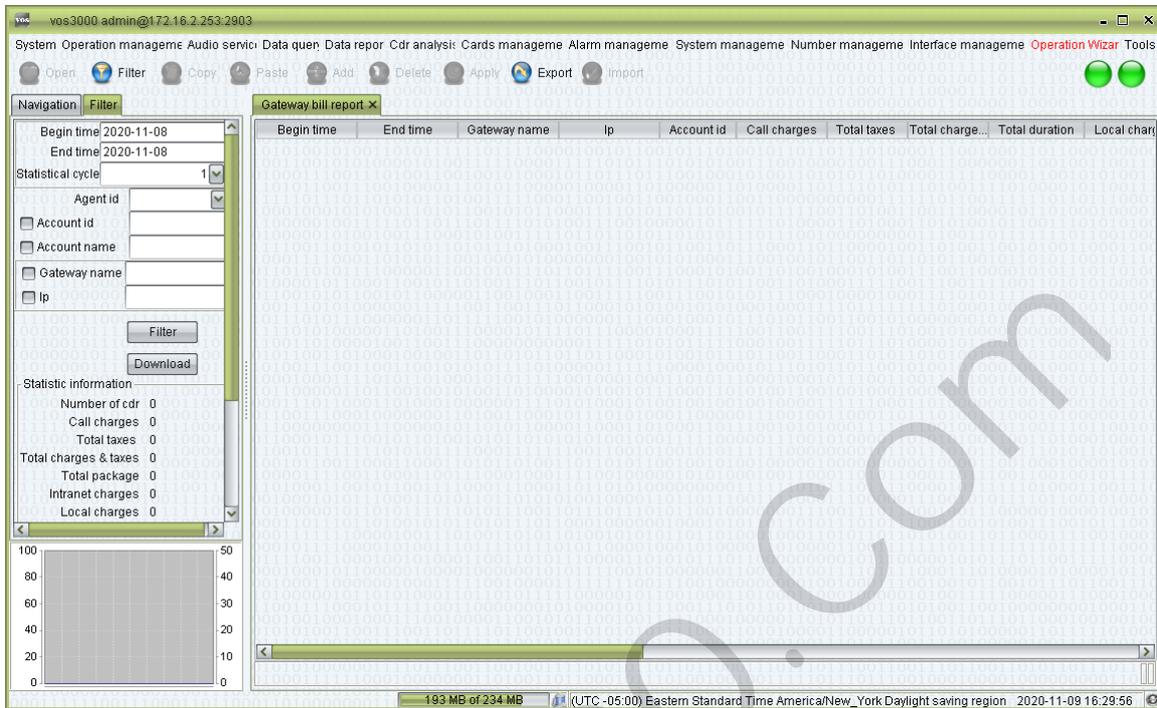
- Begin time
- End time
- Account id: the number of the account being displayed.
- Account name: the name of the account being displayed.
- Call charges: the total amount of charges.
- Total taxes
- Total charges & taxes
- Total duration: the total amount of session time for all calls.
- Local charges: the amount charged for local calls.
- Local duration: the amount of session time for local calls.
- Domestic charges: the amount charged for national calls.
- Domestic duration: the amount of session time for national calls.
- International charges: the amount charged for international calls.
- International duration: the amount of session time for international calls.
- Intranet charges: the amount charged for net calls.

- Intranet duration: the amount of session time for net calls.
- Total package: the total consumption of gift amount.
- Package duration: the total consumption of free duration.
- Number of cdr: the total number of phone records.

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2.8.1.2 Gateway Bill Report

This function is used to query mapping gateway's consumption report.



How to Start

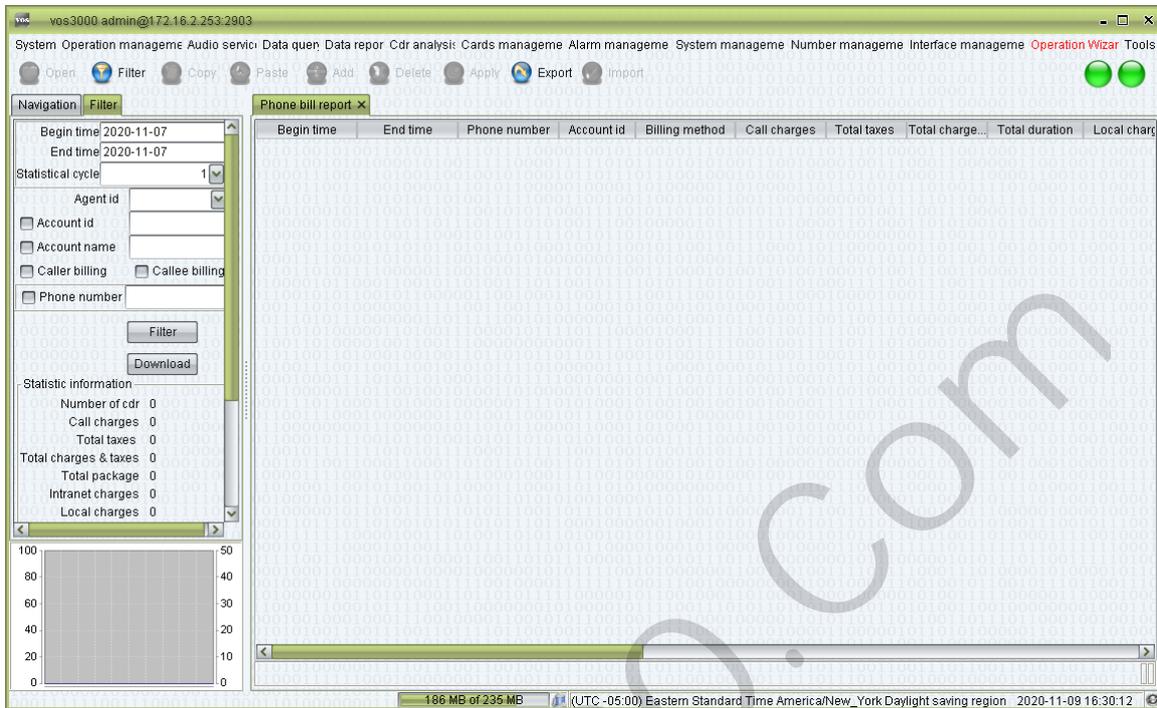
- Double-click “Navigation > Data report > Bill report > Gateway bill report”

Table Items

- Gateway name: the unique name of the device, used for the authentication of dynamic gateways. For static gateways (usually relay gateways), the only requirement is that their ids do not conflict with one another.
- Ip: the IP address of the gateway.
- Please refer to the descriptions in “Revenue detail report” for further instructions.

2.8.1.3 Phone Bill Report

This function is used to query phone's consumption report.



How to Start

- Double-click “Navigation > Data report > Bill report > Phone bill report”

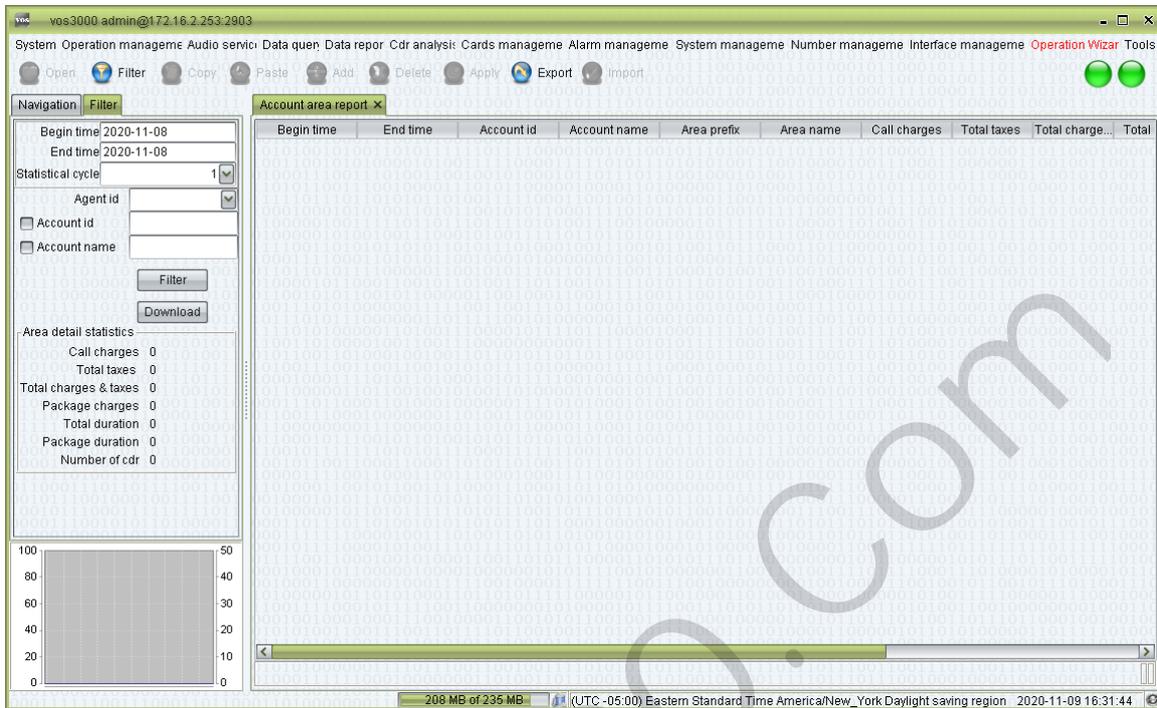
Table Items

See the descriptions in “Revenue detail report”.

- Phone number: the number used as caller id and the called number for the terminal.
- Account id: the number of the account that the phone belongs to.
- Billing method: whether the caller or the called is charged.

2.8.1.4 Account Area Report

This function is used to query area consumption of account report.



How to Start

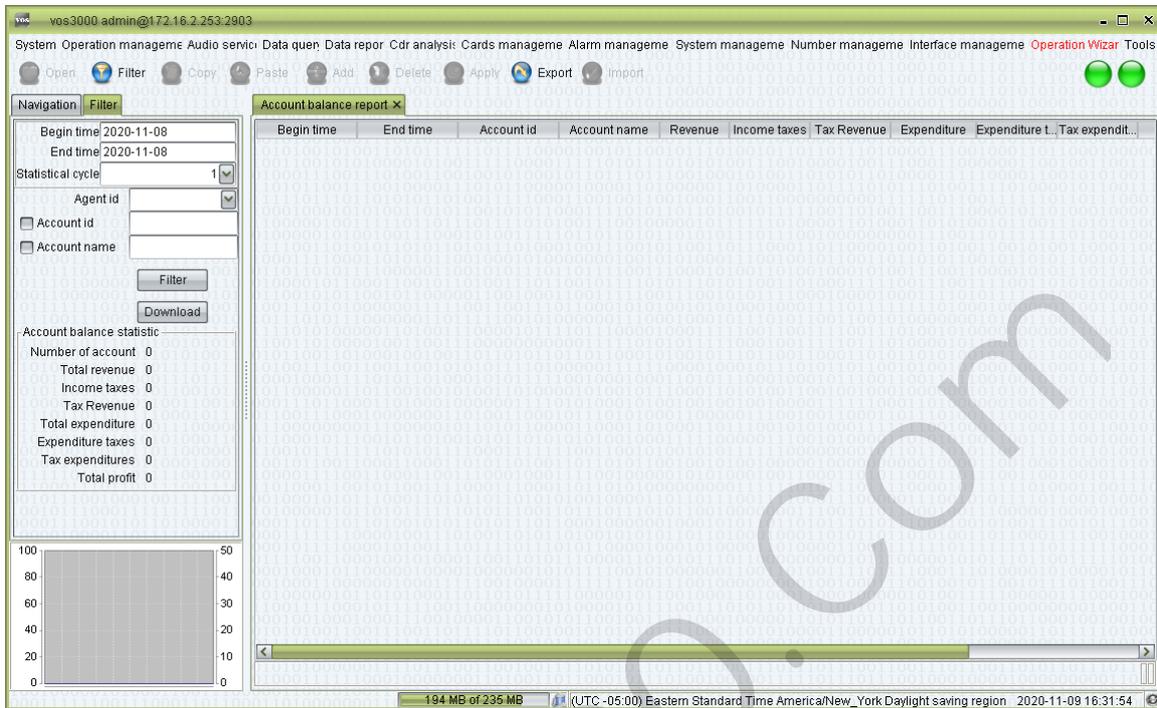
- Double-click “Navigation > Data report > Bill report > Account area report”

Table Items

See the descriptions in “Revenue detail report”.

2.8.1.5 Account Balance Report

This function is used to query account and agent’s revenue and expenditure report.



How to Start

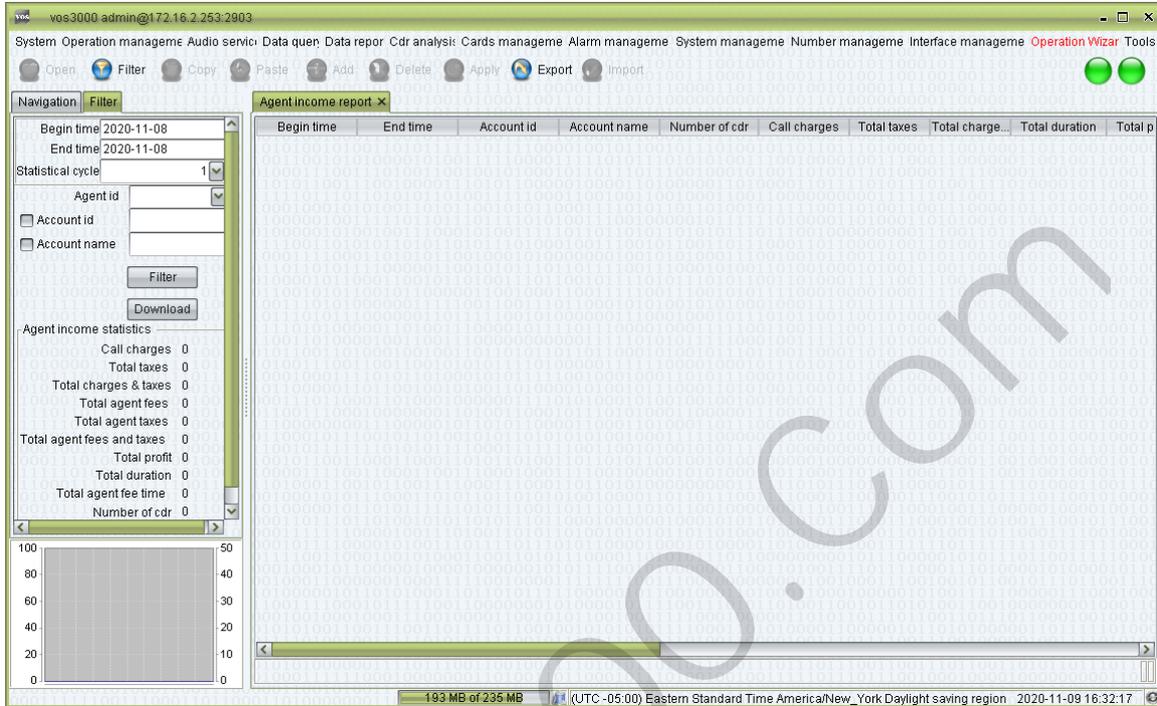
- Double-click “Navigation > Data report > Bill report > Account balance report”

Table Items

See the descriptions in “Revenue detail report”.

2.8.1.6 Agent Income Report

This function is used to display each sub-account under the agents account make profits for the agents. However Account Balance Report only display agents account income and expenses overall.



How to Start

- Double-click “Navigation > Data report > Bill report > Agent income report”

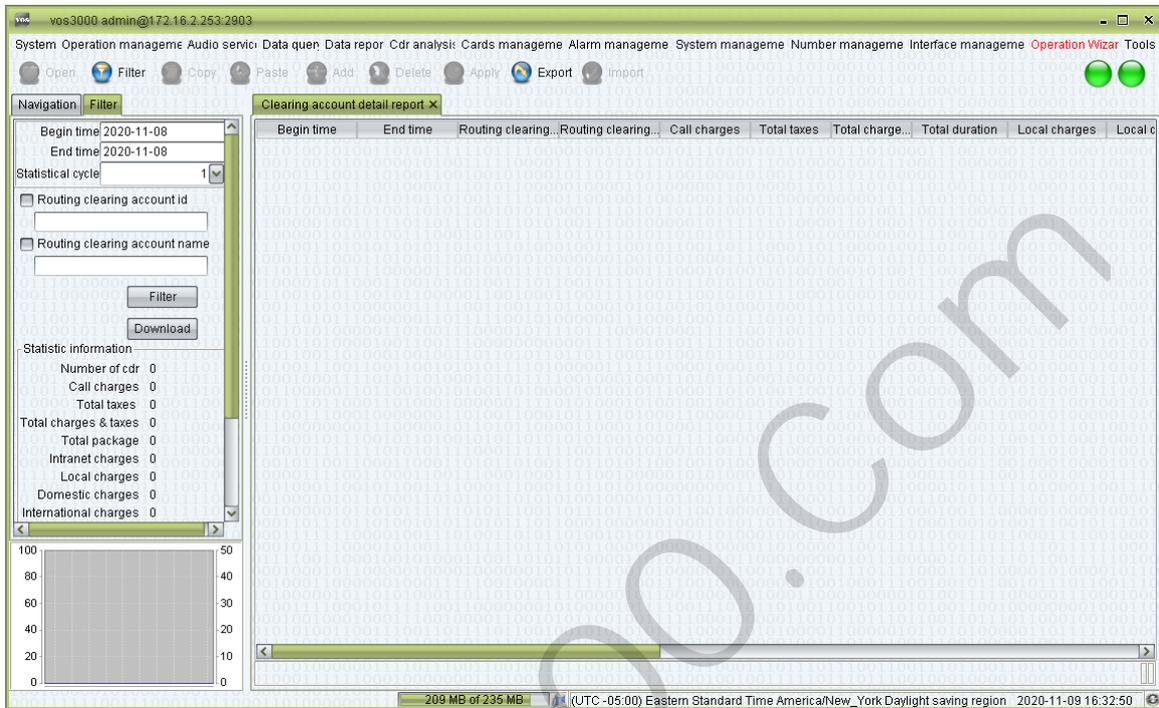
Table Items

- Total charges: the sum of the expenses generated by the agents sub-account.
- Total agent fees: the sum of agent cost for this sub-account's consumption.

2.8.2 Clearing Report

2.8.2.1 Clearing Account Detail Report

This function is used to query clearing account's consumption report.



How to Start

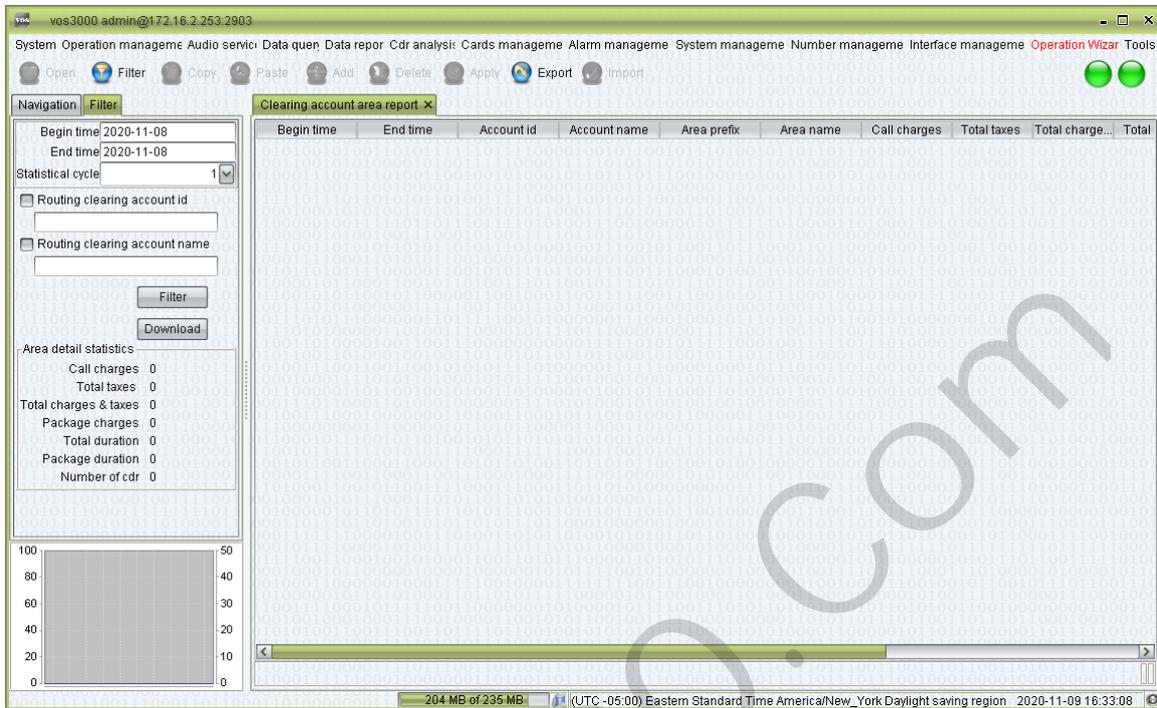
- Double-click “Navigation > Data report > Clearing report > Clearing account detail report”

Table Items

See the descriptions in “Revenue detail report”.

2.8.2.2 Clearing Account Area Report

This function is used to query clearing account's area consumption report.



How to Start

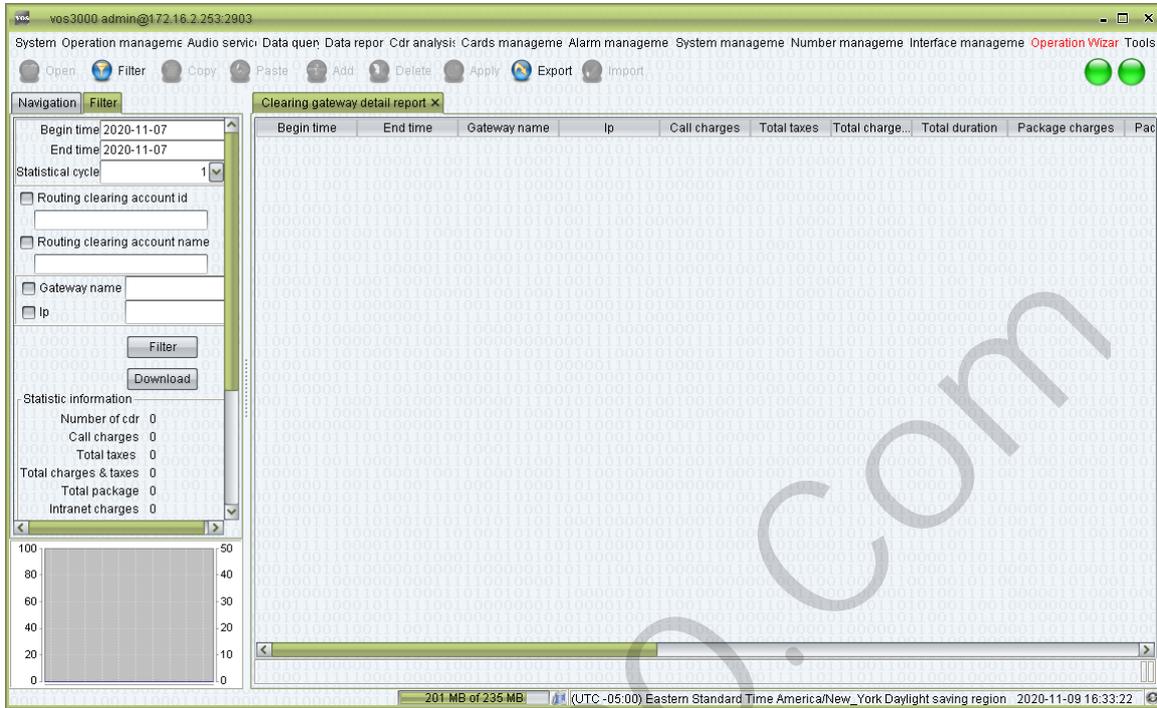
- Double-click "Navigation > Data report > Clearing report > Clearing account area report"

Table Items

See the descriptions in "Revenue detail report".

2.8.2.3 Clearing Gateway Detail Report

This function is used to query routing gateway’s consumption report.



How to Start

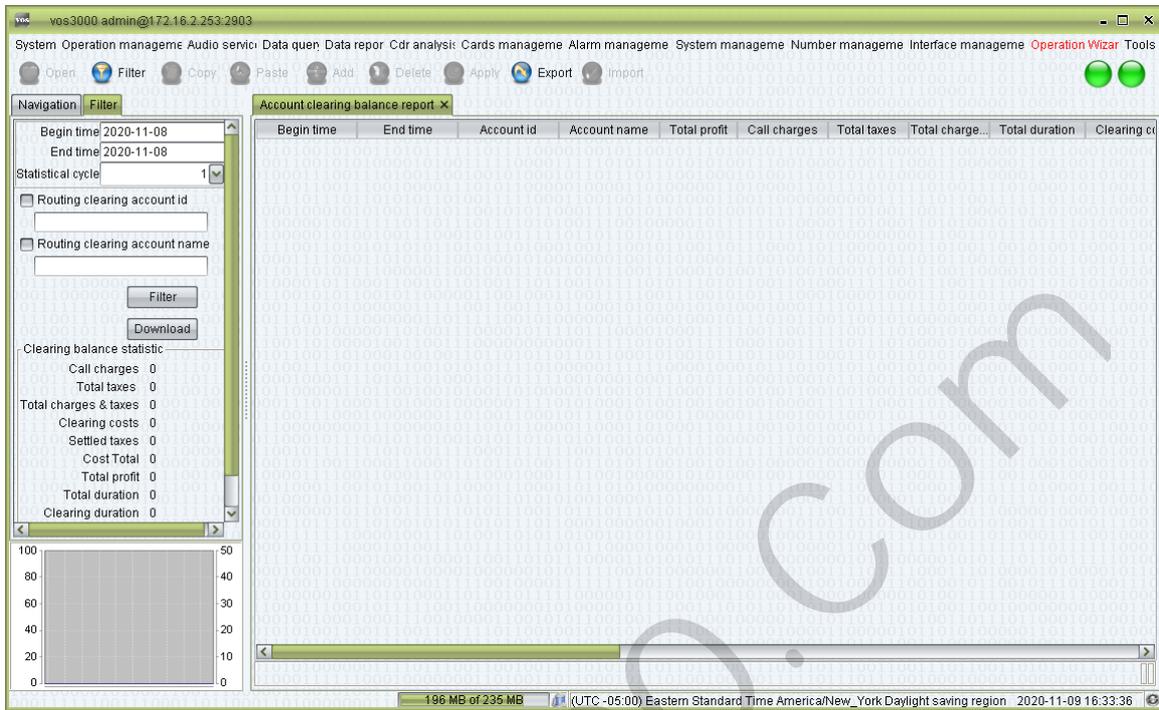
- Double-click “Navigation > Data report > Clearing report > Clearing gateway detail report”

Table Items

See the descriptions in “Revenue detail report”.

2.8.2.4 Account Clearing Balance Report

This function is used to query account's clearing consumption report.



How to Start

- Double-click “Navigation > Data report > Clearing report > Account clearing balance report”

Table Items

See the descriptions in “Revenue detail report”.

2.8.3 Analysis Report

- Total calls: total unconnected and total connected.
- Fail: unconnected calls.
- Success: calls with connect/busy/no answer/ringing signaling.



NOTE

Ringing: callee sent SIP 180 or H323 Alerting.

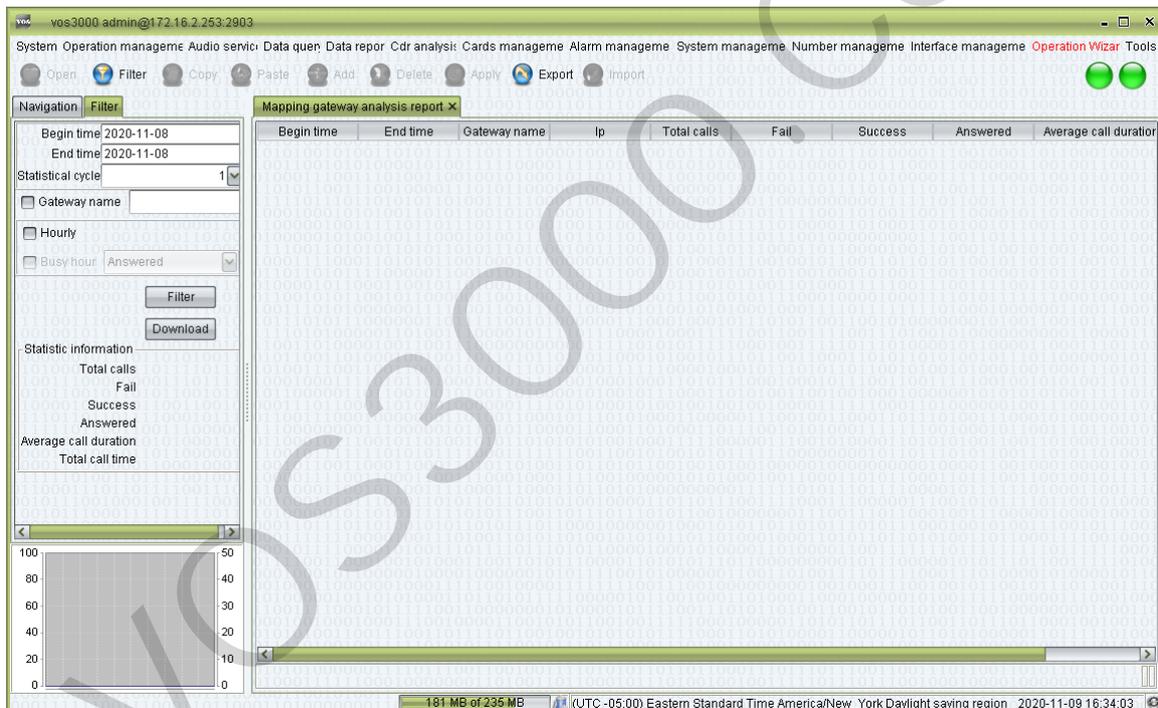
If callee is phone, sent SIP 183 with SDP or H323 CallProceeding(OLC).

If callee is routing gateway, sent SIP 183 with SDP or H323 CallProceeding(OLC) and enable SIP “Stop switch gateway after receive sdp” or H323 “Stop switch gateway after olc”.

- Answered: calls with connect signaling.
- Average call duration: average call duration.
- Total call time: total duration.

2.8.3.1 Mapping Gateway Analysis Report

This function is used to analysis mapping gateway.

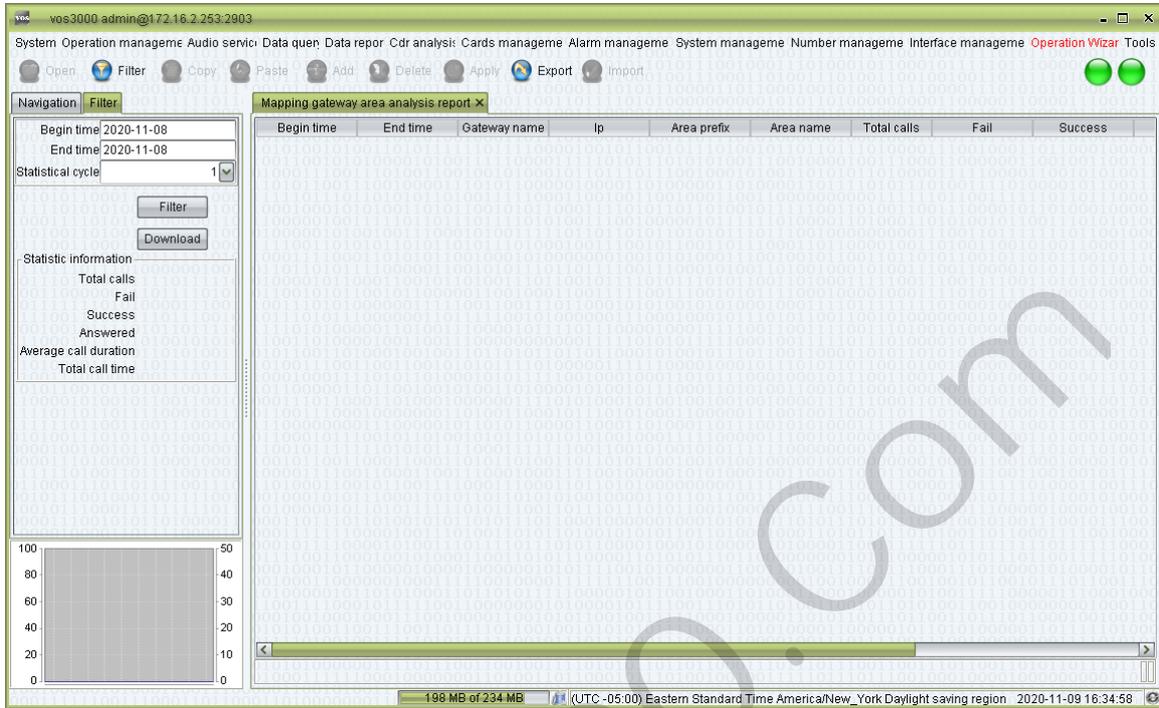


How to Start

- Double-click “Navigation > Data report > Analysis report > Mapping gateway analysis report”

2.8.3.2 Mapping Gateway Area Analysis Report

This function is used to analysis mapping gateway area.

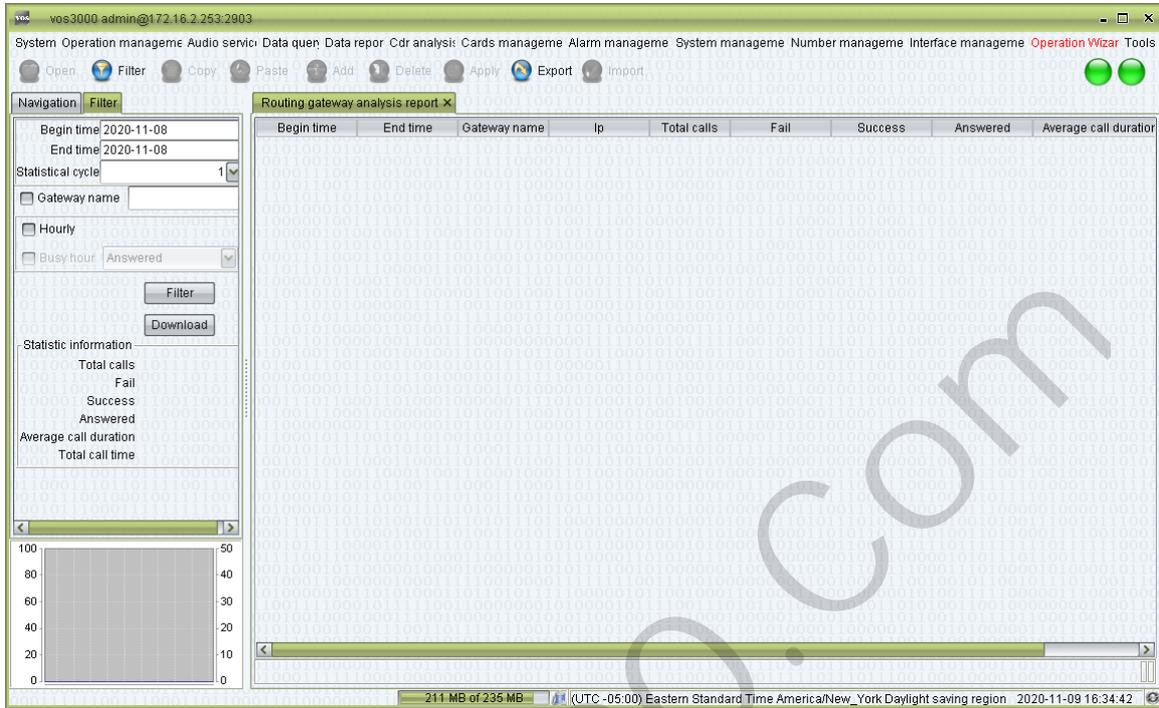


How to Start

- Double-click “Navigation > Data report > Analysis report > Mapping gateway area analysis report”

2.8.3.3 Routing Gateway Analysis Report

This function is used to analysis routing gateway.

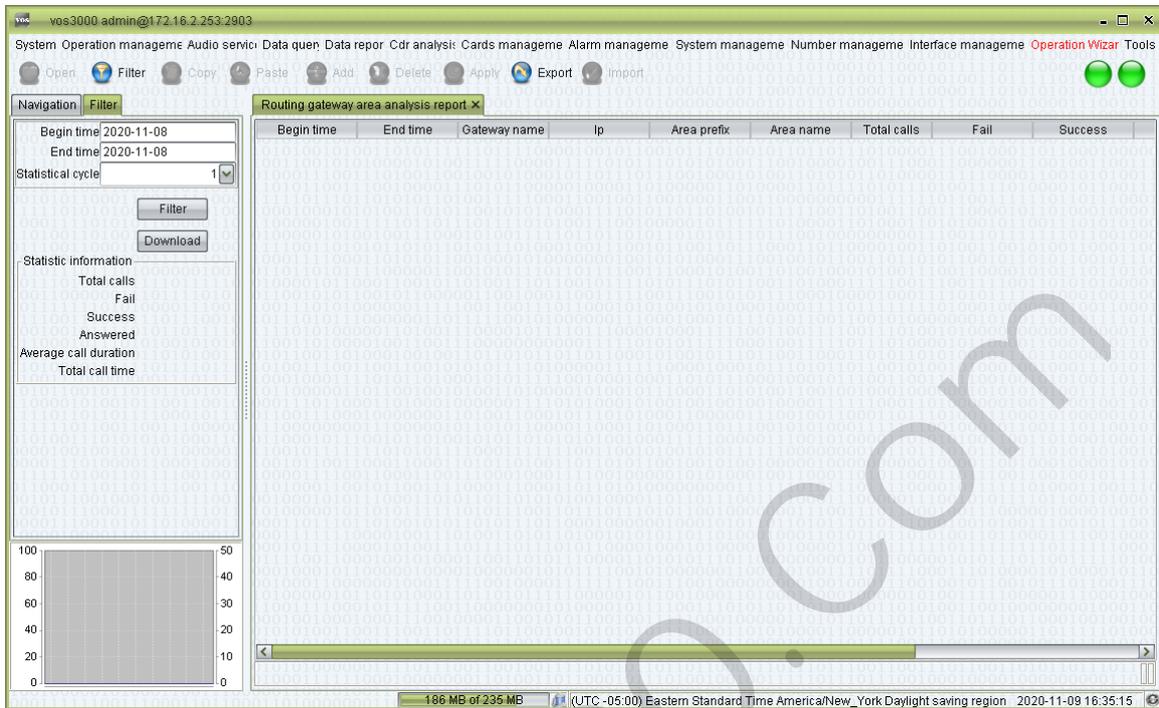


How to Start

- Double-click “Navigation > Data report > Analysis report > Routing gateway analysis report”

2.8.3.4 Routing Gateway Area Analysis Report

This function is used to analysis routing gateway area.

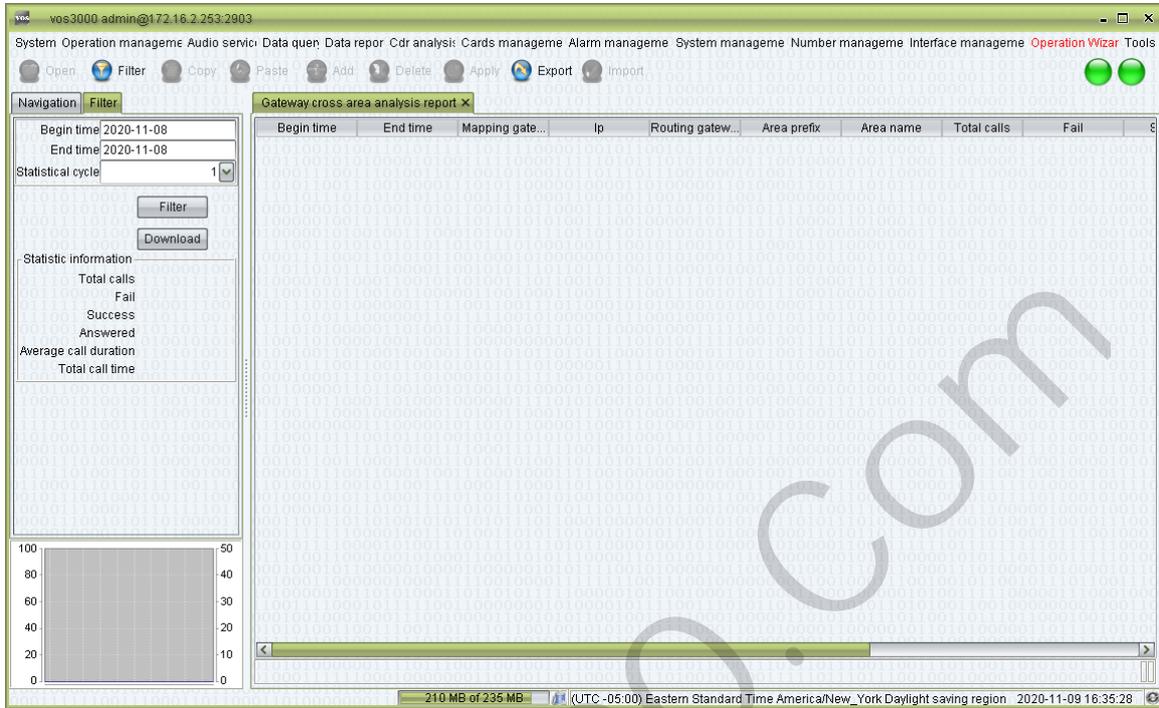


How to Start

- Double-click “Navigation > Data report > Analysis report > Routing gateway area analysis report”

2.8.3.5 Gateway Cross Area Analysis Report

This function is used to analysis mapping gateway cross routing gateway.

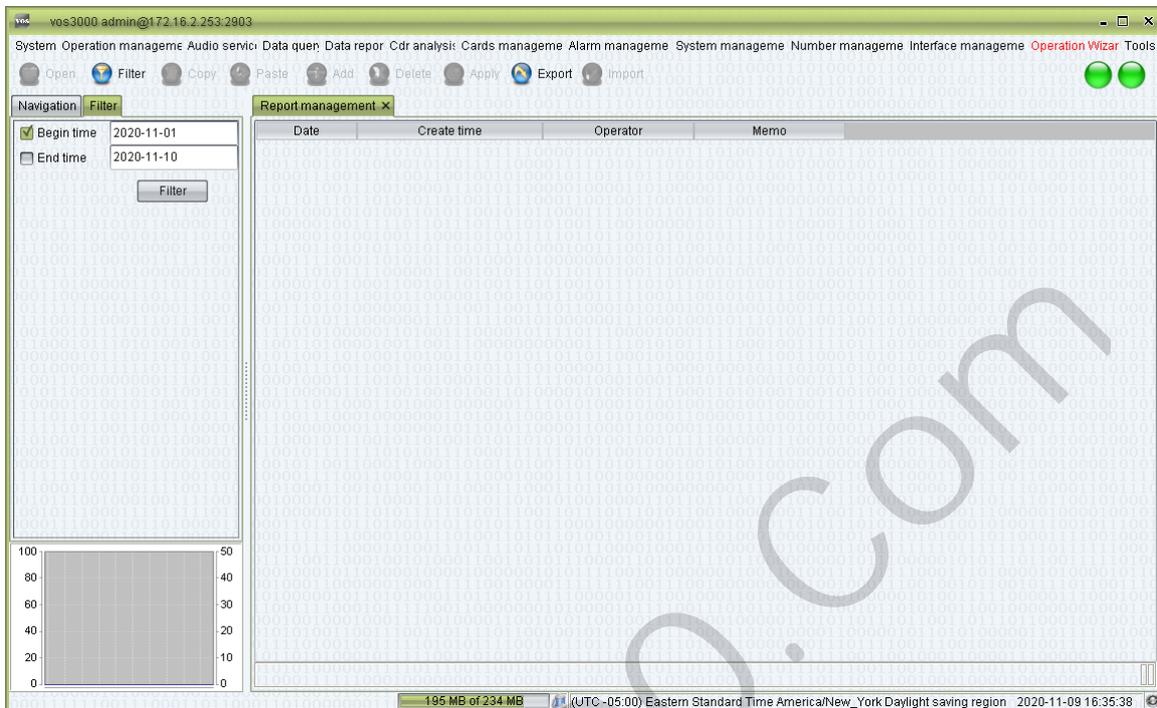


How to Start

- Double-click “Navigation > Data report > Analysis report > Gateway cross area analysis report”

2.8.4 Report Management

This function is used to manage reports.



How to Start

- Double-click “Navigation > Data report > Report management”

Table Items

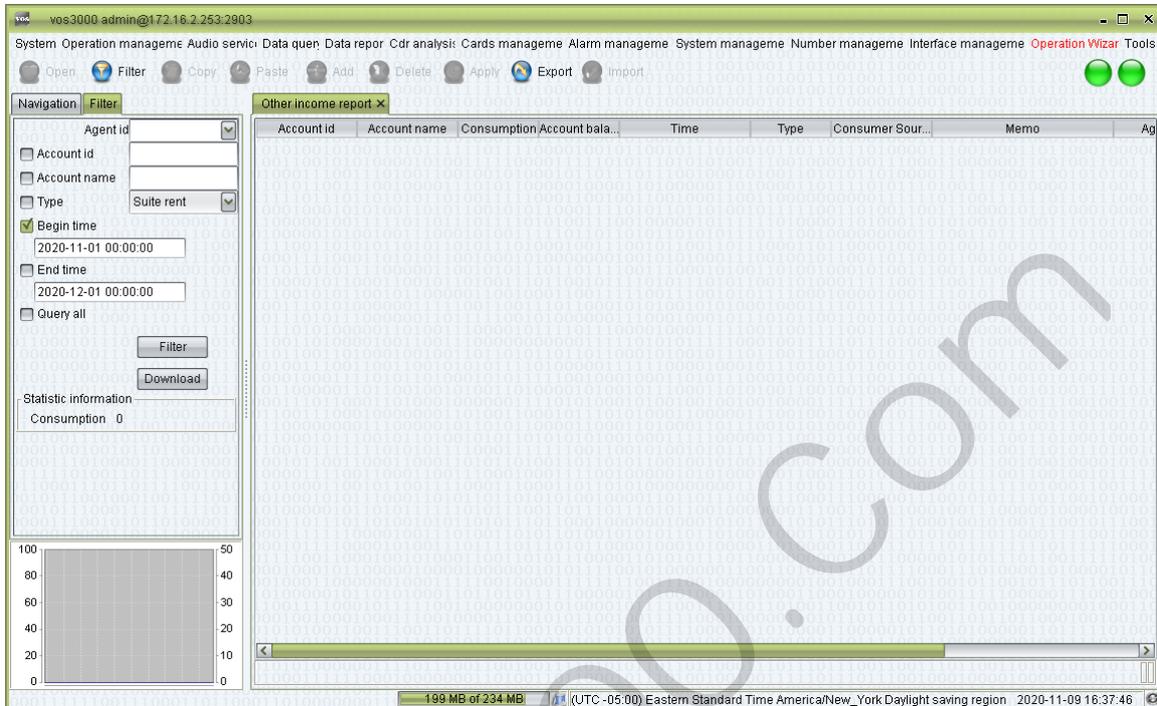
- Date: date of records in the report.
- Create time: the date when the report is generated.
- Operator: the operator that generated the report.
- Memo: items included in the report.

Right-Click Menu

- Generate all reports
- Generate special report

2.8.5 Other Income Report

This function is used to query income from suite rent, phone month rent, phone under consumption, suite under consumption.



How to Start

- Double-click “Navigation > Data report > Other income report”

Table Items

- Account id
- Account name
- Consumption
- Account balance
- Time: deduction time
- Type: suite rent, phone month rent, phone under consumption, suite under consumption.
- Consumer Source
- Memo
- Agent id
- Agent name
- Serial number

Right-Click Menu

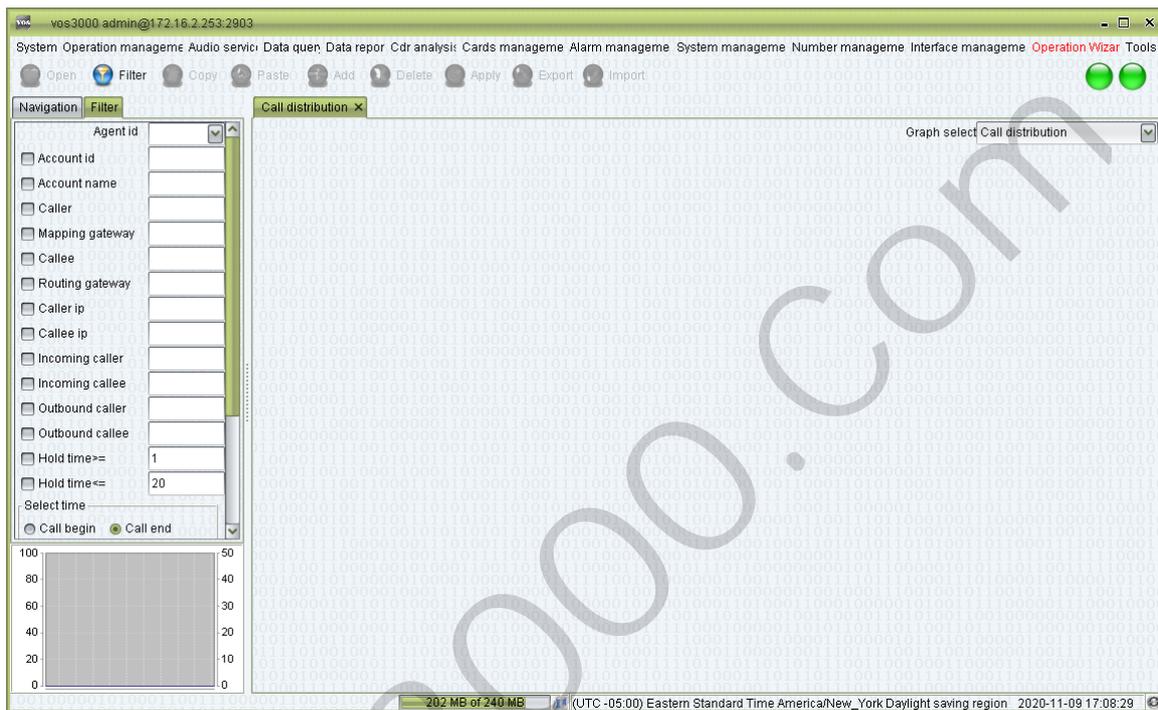
- Total up

2.9 CDR Analysis

See the descriptions in “Analysis report”.

2.9.1 Call Distribution

This function is used to show call distribution of 24 hours.



How to Start

- Double-click “Navigation > CDR analysis > Call distribution”

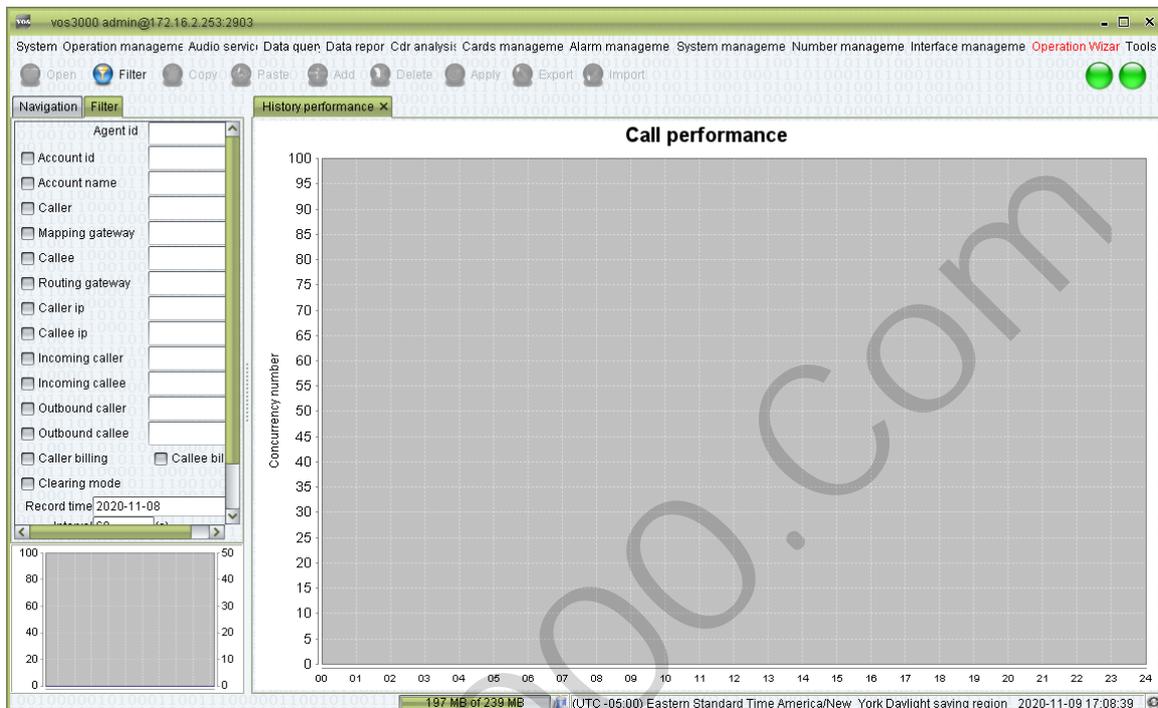
2.9.2 Historical Performance

This function is used to show the concurrent calls on specified date.



NOTE

Unsuccessful calls are not counted here, so the number shown in the chart will be slight lower than that in reality.



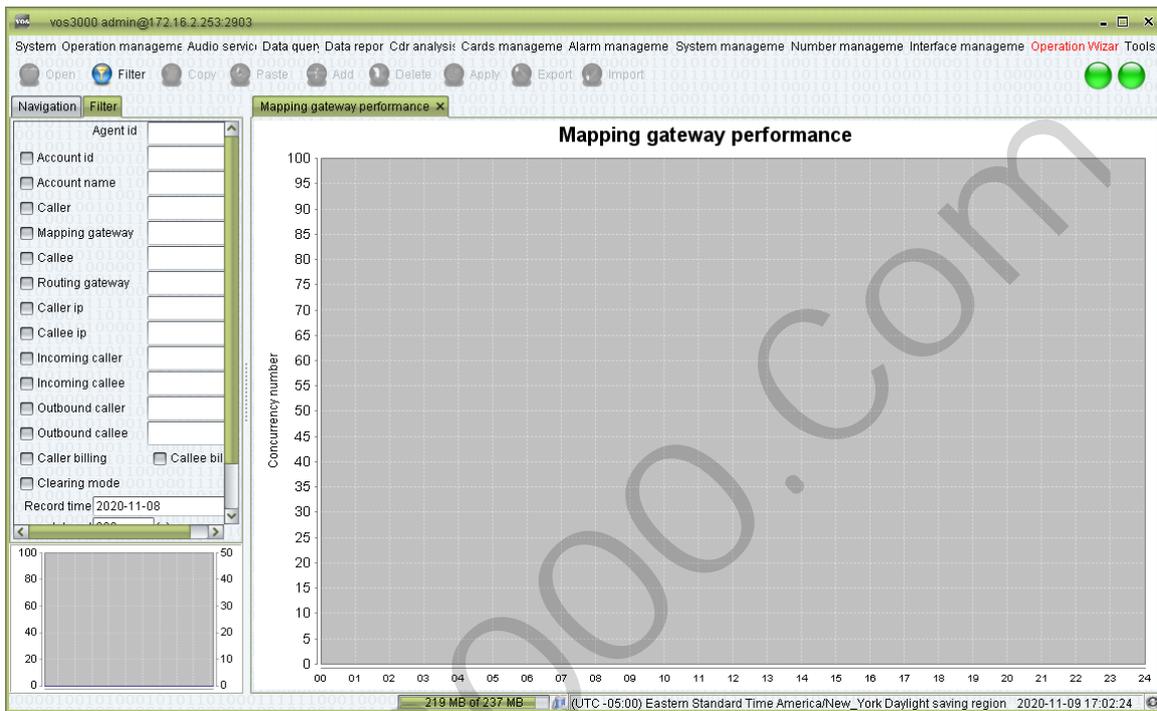
How to Start

- Double-click “Navigation > CDR analysis > Historical performance”

2.9.3 Mapping gateway Analysis

2.9.3.1 Mapping gateway performance

The performance of mapping gateway can count the number of concurrent calls in any day, and only the gateway data with the highest concurrency is displayed. The number of gateways displayed is determined by "system parameter>SERVER_DISPLAY_CHART_GATEWAY_SIZE".



How to Start

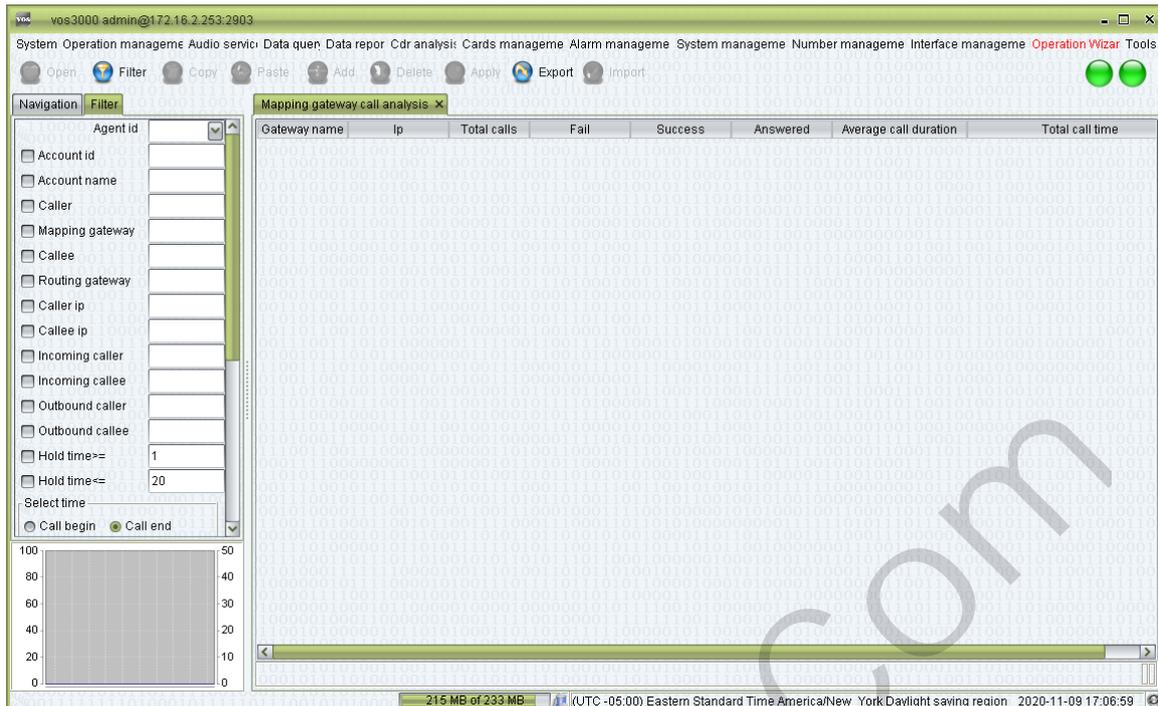
- Double-click “Navigation > CDR analysis > Mapping gateway analysis > Mapping gateway performance”

Table Items

- Longitudinal axis: concurrent quantity.
- Transverse axis: hours of the day.

2.9.3.2 Mapping gateway call analysis

This function is used to analyze the connection of the mapping gateway.



How to Start

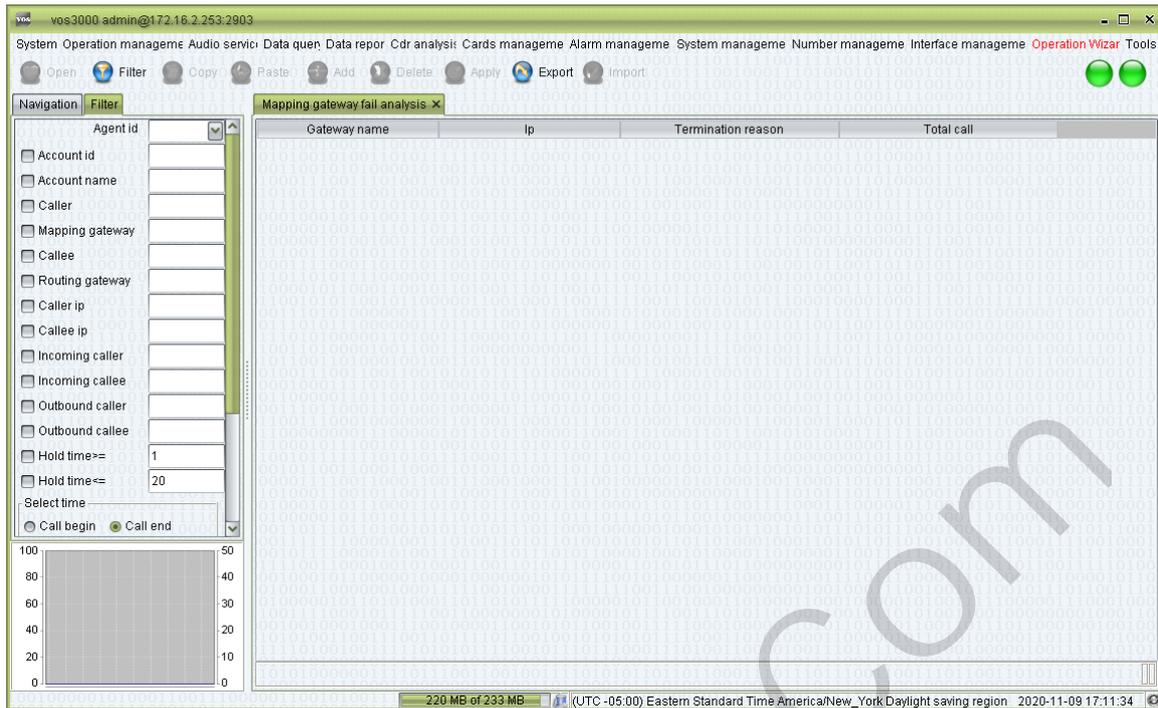
- Double-click “Navigation > CDR analysis > Mapping gateway analysis > Mapping gateway call analysis ”

Table Items

- Gateway name: it corresponds to "Mapping gateway" (when the "analysis method" in the filter condition is "caller") or "Routing gateway" (when the "analysis method" in the filter condition is "called > gateway name").
- Ip: the bill generates the calling or called IP address from (when the "analysis method" in the filter condition determines).

2.9.3.3 Mapping gateway fail analysis

Interrupt analysis can complete the analysis of the interruption of the gateway. It can analyze the interruption of the calling gateway or the called gateway respectively, and obtain the percentage of various interruption reasons.



How to Start

- Double-click “Navigation > CDR analysis > Mapping gateway analysis > Mapping gateway fail analysis”

Table Items

- Gateway name: it corresponds to "Mapping gateway" (when the "analysis method" in the filter condition is "caller") or "Routing gateway" (when the "analysis method" in the filter condition is "called > gateway name").
- Ip: the bill generates the calling or called IP address from (when the "analysis method" in the filter condition determines).

2.9.3.4 Mapping gateway call analysis daily

The time interval connection analysis can be used to analyze the daily bill in stages, sampling every 15 minutes.

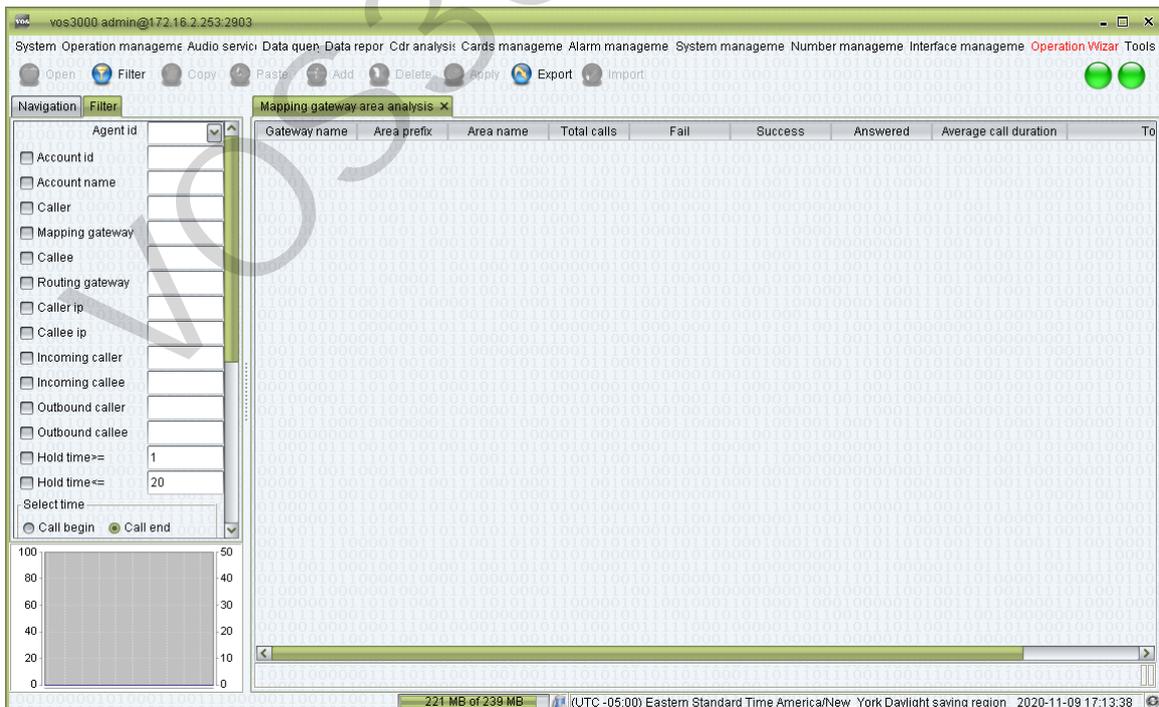


How to Start

- Double-click “Navigation > CDR analysis > Mapping gateway analysis > Mapping gateway call analysis daily”

2.9.3.5 Mapping gateway area analysis

This function displays the call analysis from the mapping gateway to each area in any time period. Refer to the "mapping gateway connection analysis report".



How to Start

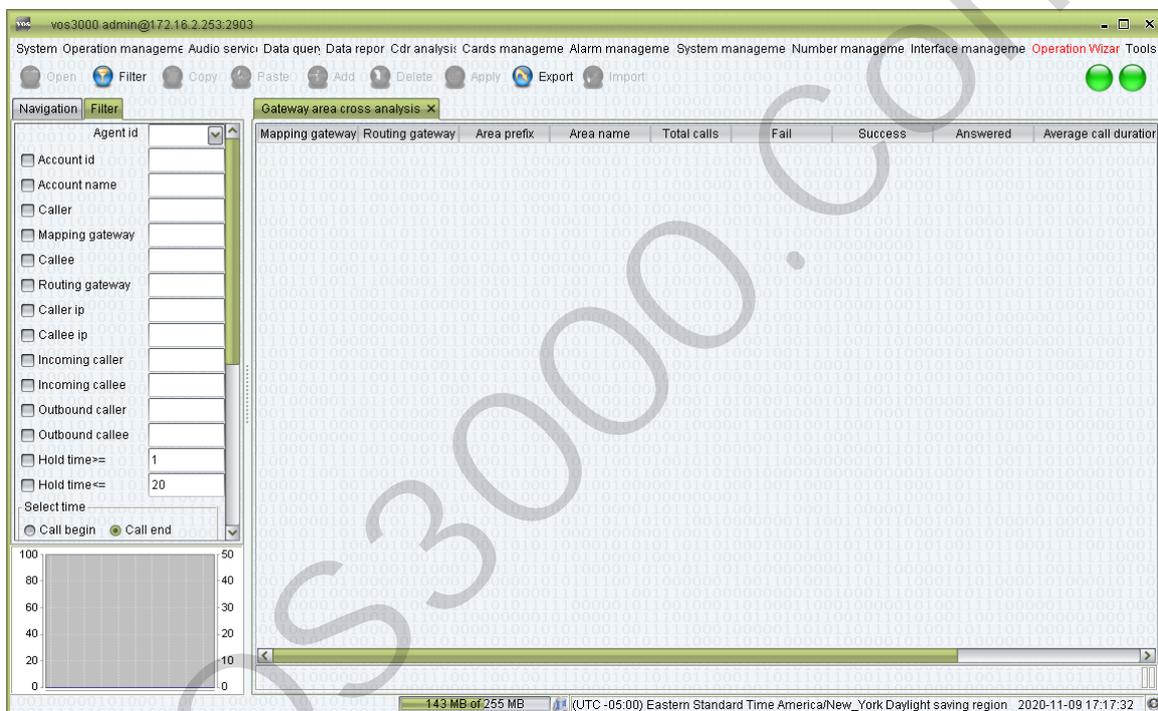
- Double-click “Navigation > CDR analysis > Mapping gateway analysis > Mapping gateway area analysis ”

Table Items

- Area prefix: “Region prefix” configured for the corresponding rate when the bill is generated.
- Area name: “Region name” corresponding to the “region prefix” setting configured by “region information”.

2.9.3.6 Gateway area cross analysis

This function displays the call analysis of mapping gateway in any time period to each region through different routing gateways. Refer to “gateway cross area analysis report”.



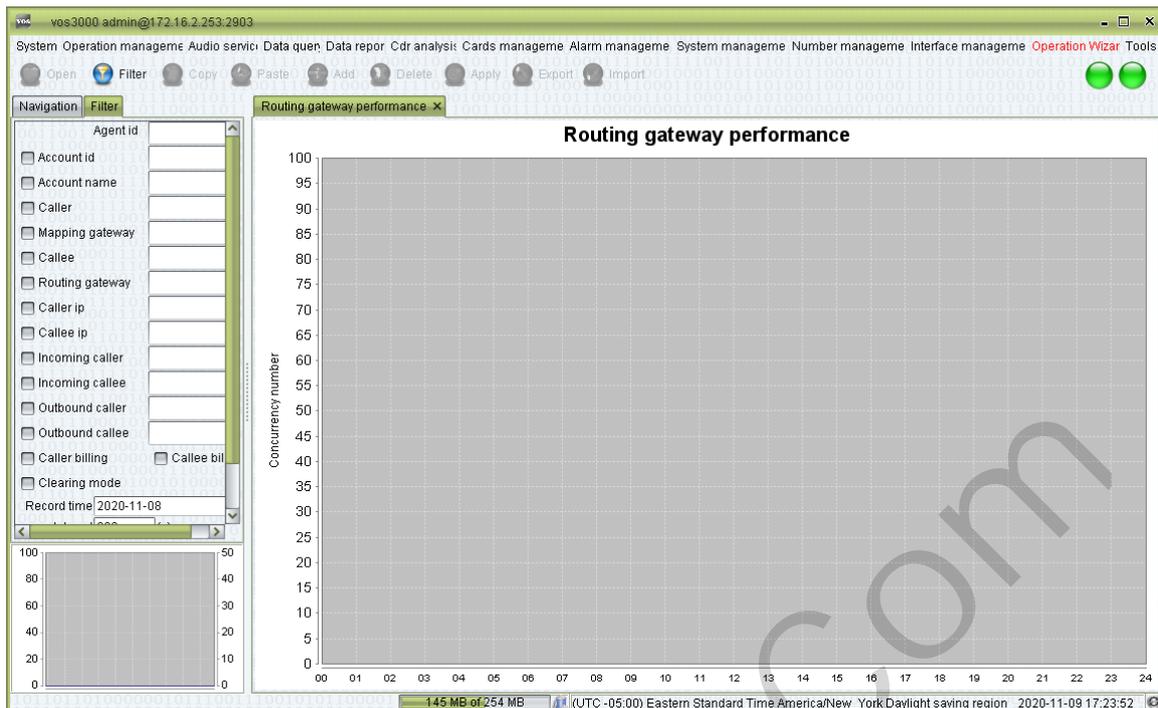
How to Start

- Double-click “Navigation > CDR analysis > Mapping gateway analysis > gateway area cross analysis ”

2.9.4 Routing gateway Analysis

2.9.4.1 Routing gateway performance

The performance of routing gateway can count the number of concurrent calls in any day, and only the gateway data with the highest concurrency is displayed. The number of gateways displayed is determined by "system parameter>SERVER_DISPLAY_CHART_GATEWAY_SIZE".



How to Start

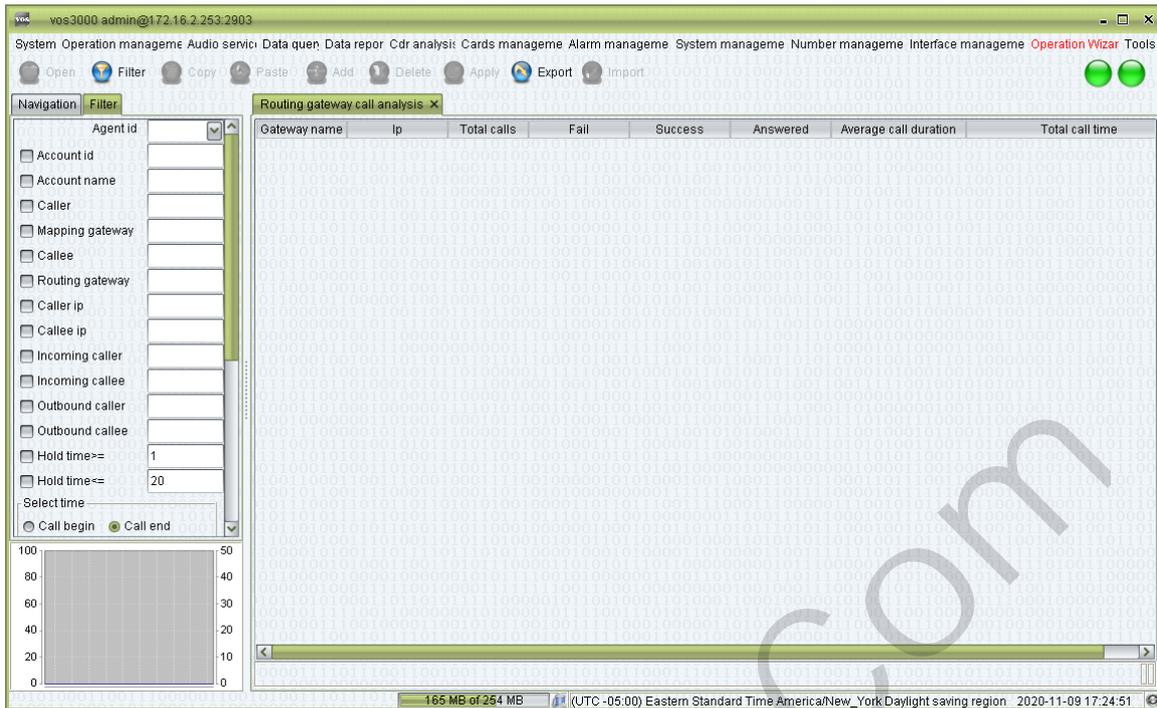
- Double-click “Navigation > CDR analysis > Routing gateway analysis > Routing gateway performance”

Table Items

- Longitudinal axis: concurrent quantity.
- Transverse axis: hours of the day.

2.9.4.2 Routing gateway call analysis

This function is used to analyze the connection of the routing gateway.



How to Start

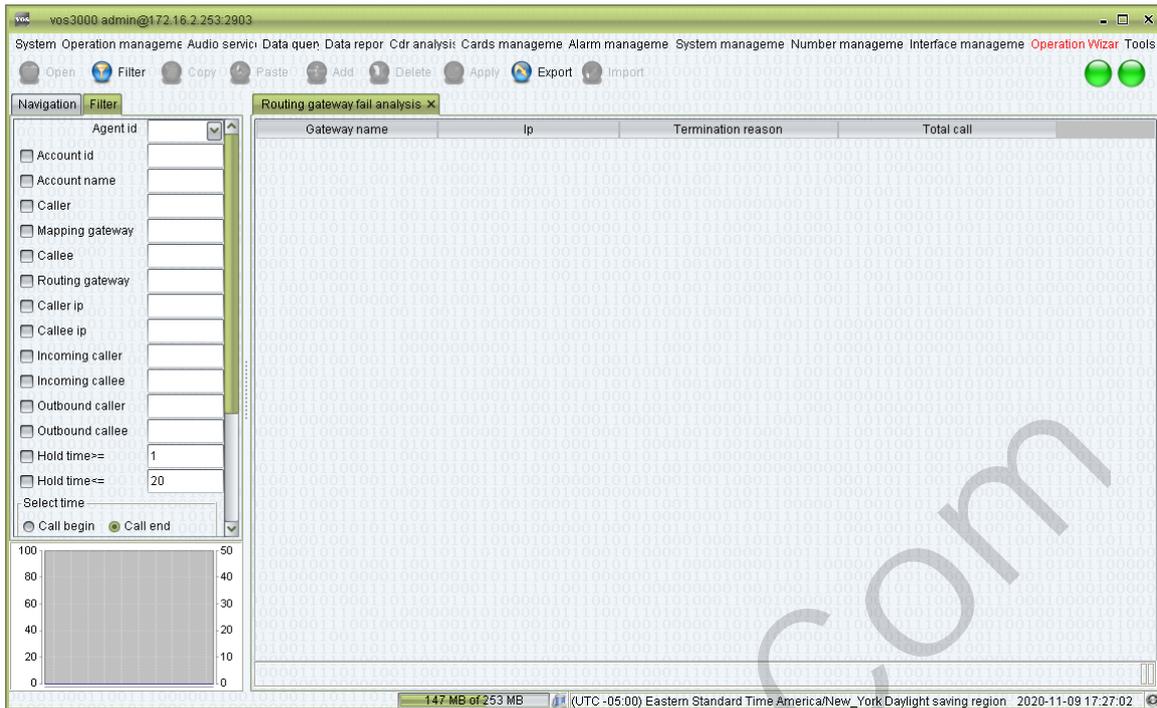
- Double-click “Navigation > CDR analysis > Routing gateway analysis > Routing gateway call analysis ”

Table Items

- Gateway name: it corresponds to "Mapping gateway" (when the "analysis method" in the filter condition is "caller") or "Routing gateway" (when the "analysis method" in the filter condition is "called > gateway name").
- Ip: the bill generates the calling or called IP address from (when the "analysis method" in the filter condition determines).

2.9.4.3 Routing gateway fail analysis

Interrupt analysis can complete the analysis of the interruption of the gateway. It can analyze the interruption of the calling gateway or the called gateway respectively, and obtain the percentage of various interruption reasons.



How to Start

- Double-click “Navigation > CDR analysis > Routing gateway analysis > Routing gateway fail analysis”

Table Items

- Gateway name: it corresponds to "Mapping gateway" (when the "analysis method" in the filter condition is "caller") or "Routing gateway" (when the "analysis method" in the filter condition is "called > gateway name").
- Ip: the bill generates the calling or called IP address from (when the "analysis method" in the filter condition determines).

2.9.4.4 Routing gateway call analysis daily

The time interval connection analysis can be used to analyze the daily bill in stages, sampling every 15 minutes.

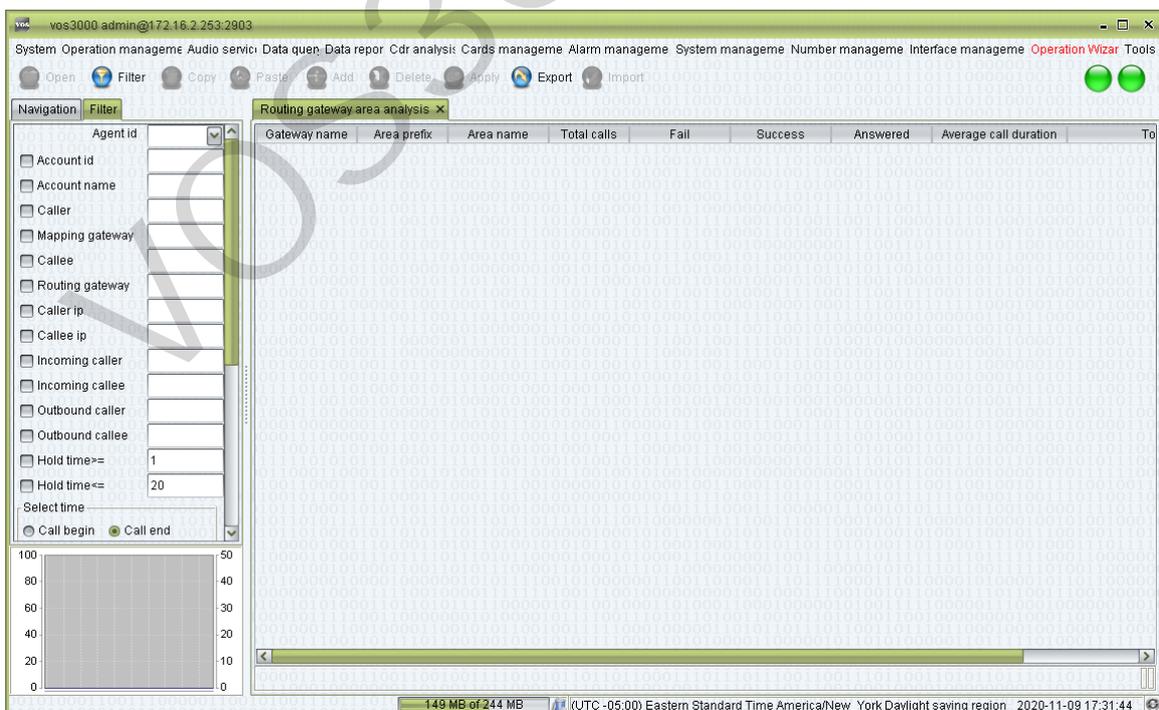


How to Start

- Double-click "Navigation > CDR analysis > Routing gateway analysis > Routing gateway call analysis daily"

2.9.4.5 Routing gateway area analysis

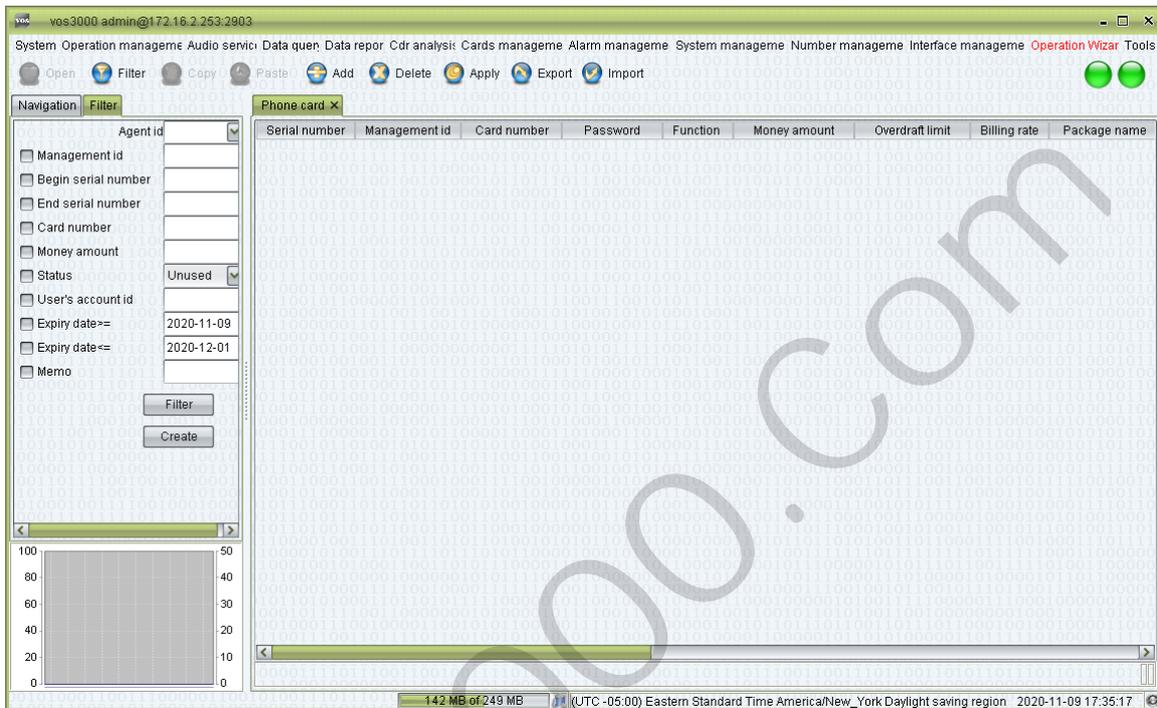
This function displays the call analysis from the routing gateway to each area in any time period. Refer to the "routing gateway connection analysis report".



2.10 Cards Management

2.10.1 Phone Card

This function is used to manage phone card.

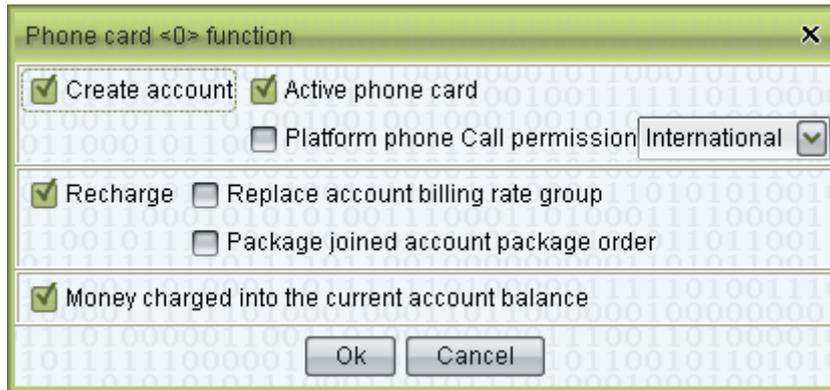


How to Start

- Double-click “Navigation > Card management > Phone card”

Table Items

- Serial number: the sequence number of activated cards. The initial number can be specified by users.
- Management id: the classification of cards can be classified according to this number.
- Card number: the unique id of a phone card. Card numbers of existing cards cannot be modified.
- Password: the password used for authentication in recharge.
- Function: functions that this card can be used for.



- Create account: this card can be used to establish an account.
 - ◆ Active phone card: when using this card create account, to do an account and active phone card.
 - ◆ Platform phone: when using this card create account, to do platform phone, also can set default authorization type.
 - Recharge: this card can be used to recharge.
 - ◆ Replace account billing rate group: when using this card recharge, the current account billing rate is replaced by the set rate card.
 - ◆ Package joined account package order: when using this card recharge, the account package will add the set package card.
 - Money charged into the current account balance: when using this card create account or recharge, the current balance of this account can increase the value of card.
 - Money amount: the amount to be recharged.
 - Overdraft limit: the amount that can be overdrawn for an account generated by this card.
- NOTE**
Negative is supported.
- Billing rate: the rate used for phone card.
 - Package name: the name of the package used for phone card.
- NOTE**
Multi packages are supported.
- Agent id: the agent account specified in account binding for phone card.
 - Lock type: “No Lock” or “Locked”.
 - Sold: manually set this feature to show if the calling card is on sale.
 - Bind number limit: when used as a calling card the number limit that allows binding.
 - Display caller id: the caller ID shown at the called end.
 - Produce time: the date when the card is created.
 - Expire time: the date of expiration.
 - Active(Days): the number of days when enabled before expiration time.
 - Enable date: the date the card was used.
 - Memo: descriptions of the card.
 - User's account id: when used, the number of the account being recharged.

- User's account name: when used, the name of the account being recharged.

Batch create phone cards

Settings

Function

- Create account
- Active phone card
- Platform phone Call permission International
- Recharge
- Replace account billing rate group
- Package joined account package order
- Money charged into the current account balance

Normal

Number of cards: Money amount:

Password mode: On Password length:

Begin card number:

Others

Begin serial number: Management id:

Card number prefix: Overdraft limit:

Billing rate: Package name:

Active(days): 0 Expire time: 2019-12-20

Bind number limit: 1 Lock type: No lock

Agent id:

Memo:

Ok Cancel

See “Phone card”.

- Number of cards: the number of cards to be created.
- Money amount: the amount to be recharged.
- Password mode: whether the cards being created have passwords.
- Password length: the length of the passwords. Passwords are generated automatically by the system.
- Begin card number: the initial card number of the cards.
- Begin serial number: the initial sequence number of the cards. The last sequence number will be automatically determined according to the number of card to be created.



NOTE

If the initial sequence number is left blank, it will be set to the number that is one plus the largest sequence number activated up to now.

- Management id
- Card number prefix: the prefix number of the cards being created.
- Overdraft limit

- Billing rate
- Package name
- Active(days): how many days will account's expiry date be extended after charging.
- Expire time
- Bind number limit
- Lock type
- Agent id: the agent account specified in account binding for phone cards.

 **NOTE**

This card can only be used for the agent's sub-account when setting the agent account.

- Memo

Related Instructions

- Phone card as account cards case.
 - The active day of this phone card is 0 days, when using this card create account ,the account is valid for phone card expire time.
 - The active day of this phone card is N days,when using this card create account , the account is valid for the current time extended N*24 hours.
- Phone card as recharge cards case.
 - If the extened mode was specified as “Superposition” in “System management > System parameter > SERVER_PAY_PHONE_CARD_CUSTOMER_EXPIRE_DAY_MODE” , If phone card's active (days) is N, the account's valid time will be extended N*24 hours after recharge, from the current time.
 - If the extened mode was specified as “Standard” in “System management > System parameter > SERVER_PAY_PHONE_CARD_CUSTOMER_EXPIRE_DAY_MODE” .
 - ◆ If phone card's active (days) is 0, if the expiration date extension is specified in”System management > System parameter > SERVER_PAY_DELAY_CUSTOMER_EXPIRE_DAY”, the expiration date of an account will be re-calculated accordingly after recharge.
 - ◆ If phone card's active (days) is N, the account's valid time will be extended N*24 hours, from the current time.then the expiration date of an account will be re-calculated accordingly compared to original expired time of account.

2.11 Alarm Management

- Alarm severity: General/Minor/Major/Critical.
- Upper: upper bound which trigger the alarm.
- Lower: lower bound which trigger the alarm.
- Period: detection period.
- Email alarm: can be set by “System management > System parameter > SERVER_ALARM_ENABLE_EMAIL”.
- Email: can be set by “System management > System parameter > SERVER_ALARM_EMAIL”.

2.11.1 Alarm Settings

2.11.1.1 System Alarm

This function is used to monitor system.

Monitor device	Alarm type	Alarm severity	Upper	Period	Email alarm	Email
All	Memory	Major	90%	Default	Default	
All	Cpu	Major	90%	Default	Default	
All	Pending cdr	Major	10,000	60	Default	
All	Call duration	Minor	7,200	None	Default	
All	Clock deviation	Major	300	Default	Default	
All	Database	Critical	None	60	Default	
All	Login mac Rest.	Major	None	Default	Default	
All	Account call dur.	General	50%	Default	Default	
All	Poor performan.	Major	None	Default	Default	
All	Geofencing	Major	None	Default	Default	
All	Illegal call	Minor	600	60	Default	

How to Start

- Double-click “Navigation > Alarm management > Alarm settings > System alarm”

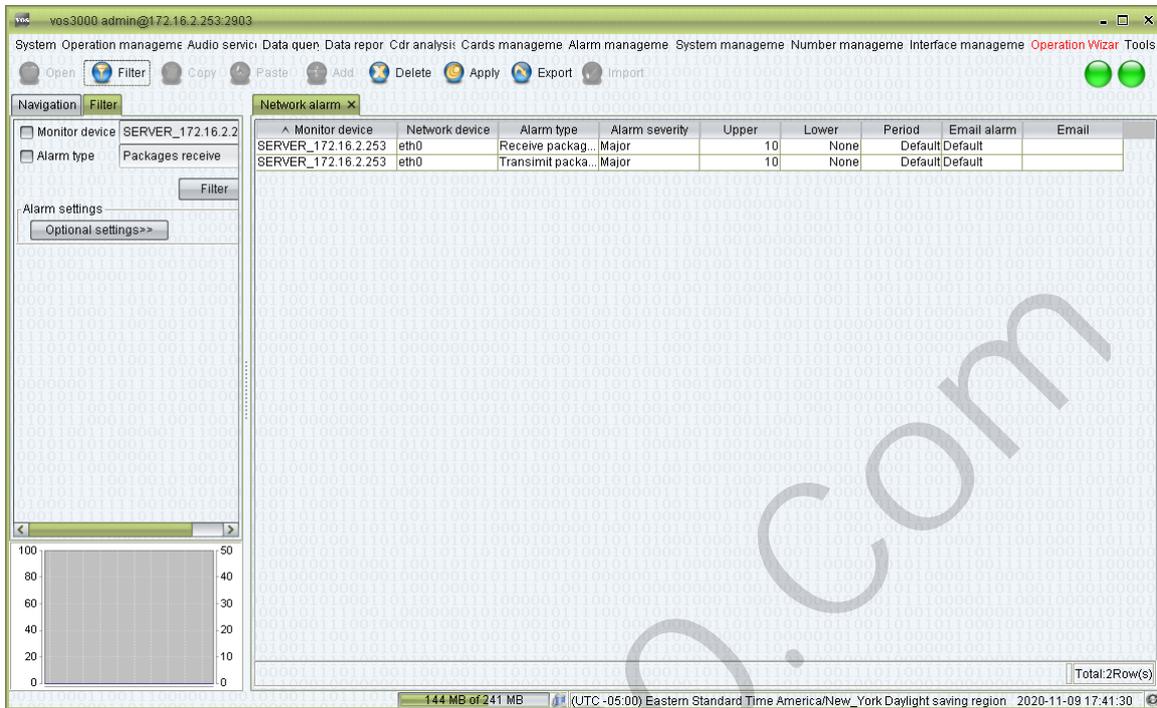
Table Items

- Monitor device
- Alarm Type:
 - Account call duration too long: alarm if the account call duration is greater than the set value.

- Call duration: alarm if the maximum time of the current call is greater than the set value.
- Clock deviation: used to monitor the time deviation between master and slave server.
- Cpu: alarm when CPU utilization is greater than set value during the detection cycle.
- Database: monitor all errors in the database except for primary key conflicts and trigger alarms.
- Host state conflict: alarm if the host state is conflict.
- Illegal call: alarm when the number of illegal calls during the detection cycle is greater than the set value.
- Login mac Restricted: when set "User management > Verify client mac" in "Trigger alarm" , the alarm is triggered if the client logs on to the mac that is inconsistent with the data in the configured client mac list.
- Master synchronize status: alarm when host synchronization stops working.
- Memory: alarm when memory usage is greater than set value during the detection cycle.
- Pending cdr: the server receives CDR and calculates CDR using an asynchronous mechanism to generate this alarm when the server is unable to process the stacked CDR in a timely manner.
- Slave synchronize status: alarm when standby synchronization stops working.
- Standby enable: this setting is valid when there is a hot standby module.

2.11.1.2 Network Alarm

This function is used to monitor network.



How to Start

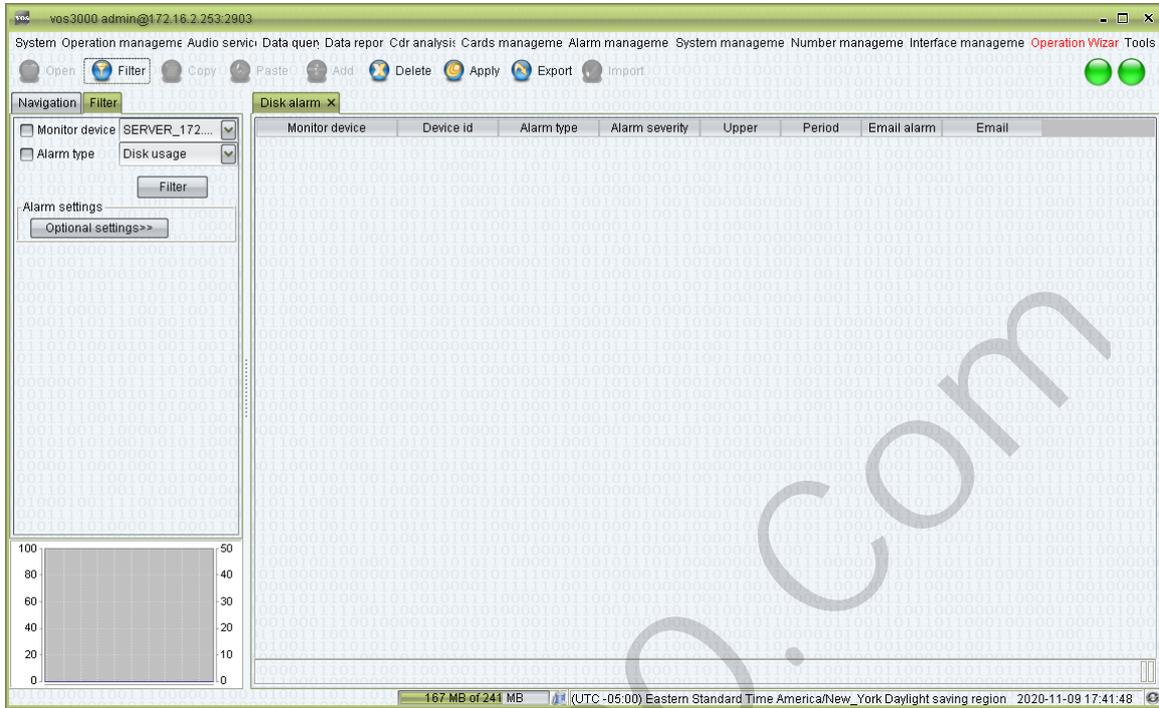
- Double-click “Navigation > Alarm management > Alarm settings > Network alarm”

Table Items

- Monitor device
- Network device
- Alarm type:
 - Bytes receive
 - Bytes transmit
 - Packages receive
 - Packages transmit
 - Receive package error
 - Transmit package error

2.11.1.3 Disk Alarm

This function is used to monitor disk.



How to Start

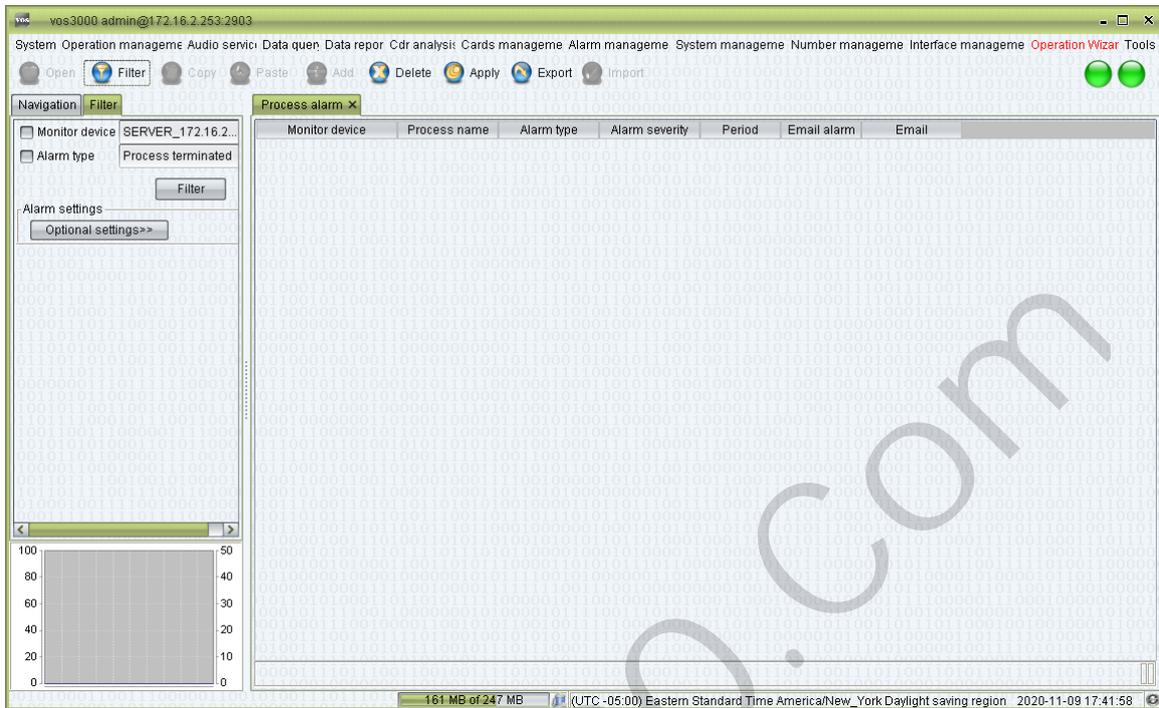
- Double-click “Navigation > Alarm management > Alarm settings > Disk alarm”

Table Items

- Monitor device
- Device id
- Alarm type:
 - Disk usage

2.11.1.4 Process alarm

This function is used to monitor process.

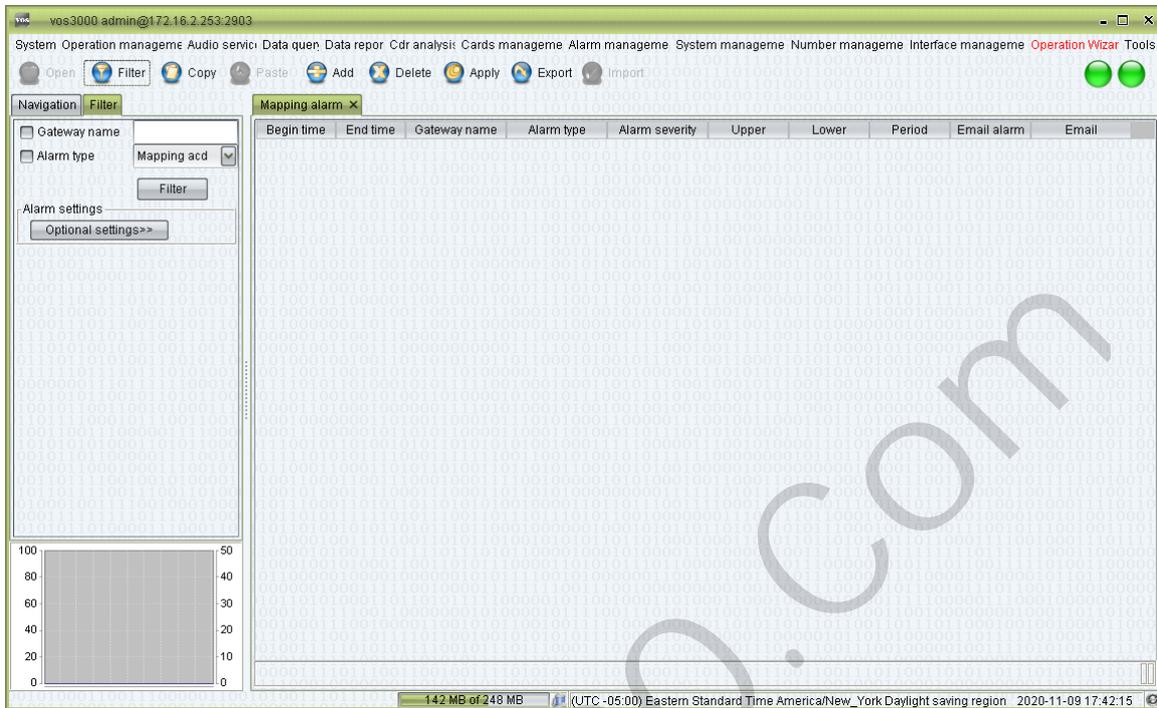


How to Start

- Double-click "Navigation > Alarm management > Alarm settings > Process alarm"

2.11.1.5 Mapping Alarm

This function is used to monitor mapping gateway.



How to Start

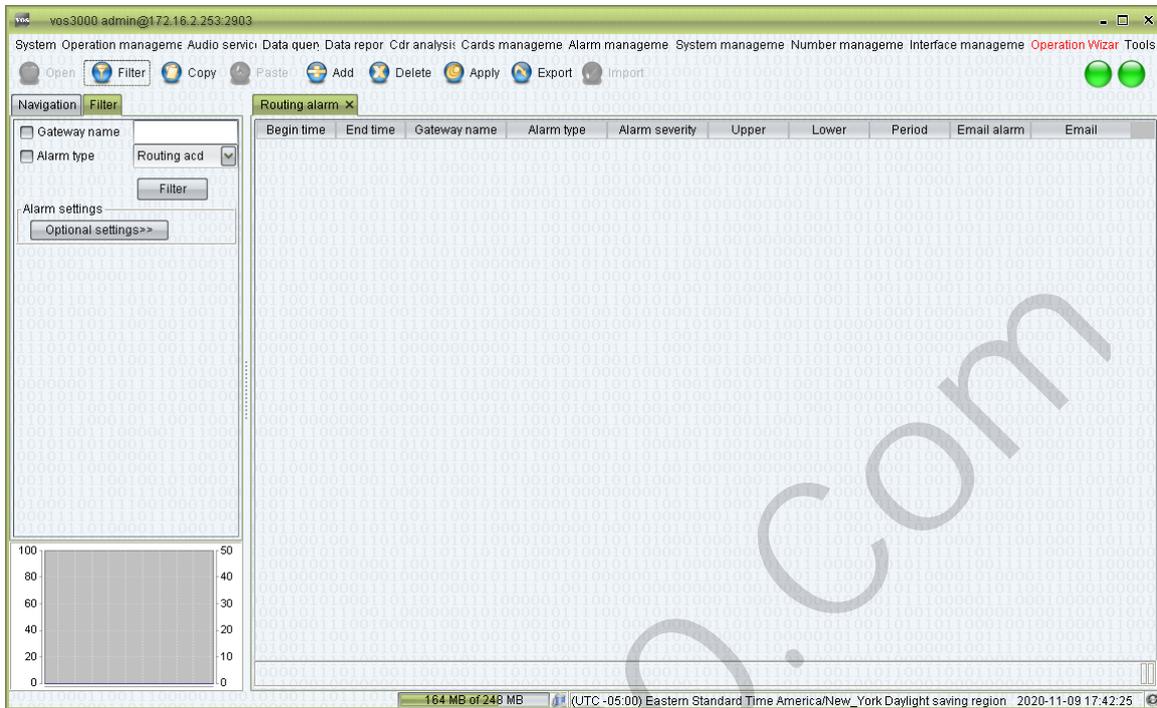
- Double-click “Navigation > Alarm management > Alarm settings > Mapping alarm”

Table Items

- Begin time
- End time
- Gateway name
- Alarm type:
 - Mapping asr
 - Mapping acd
 - Mapping concurrency decline
 - Mapping gateway concurrent rise
 - Mapping gateway call rate
 - Mapping bilateral reconciliation deviation
 - Mapping packet loss rate

2.11.1.6 Routing Alarm

This function is used to monitor routing gateway.



How to Start

- Double-click “Navigation > Alarm management > Alarm settings > Routing alarm”

Table Items

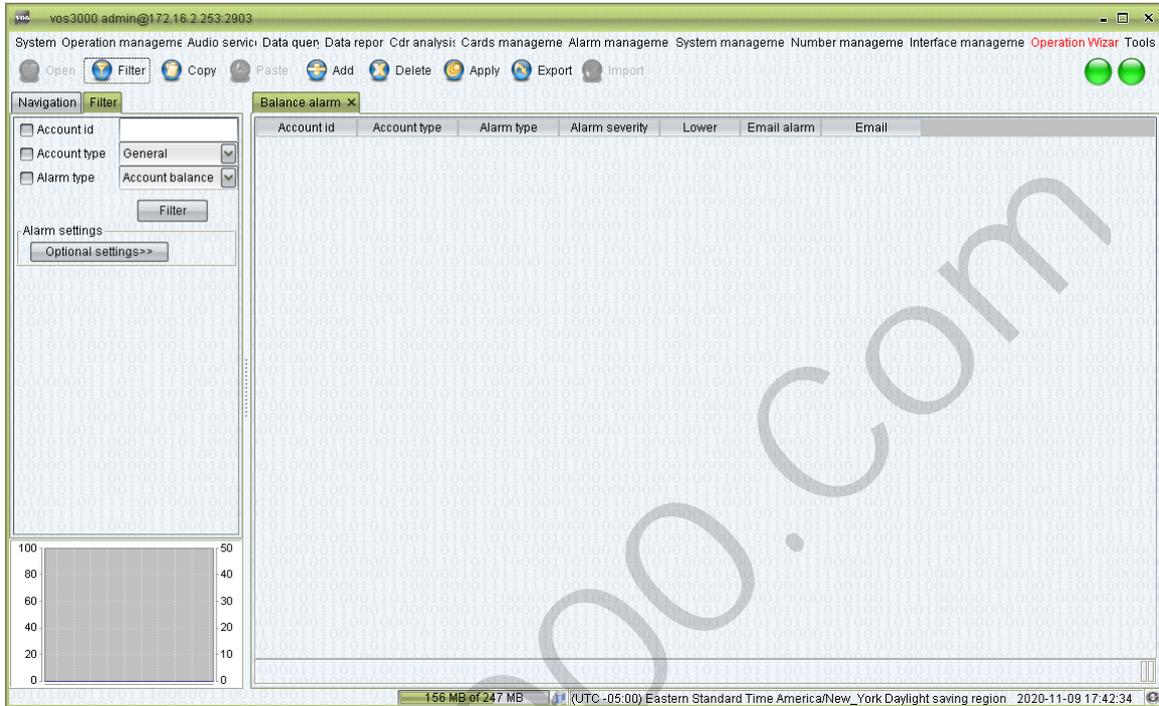
- Begin time
- End time
- Gateway name
- Alarm type:
 - Routing asr
 - Routing acd
 - Routing gateway concurrent rise
 - Routing concurrency decline
 - Routing gateway call timeout times
 - Routing bilateral reconciliation deviation
 - Routing packet loss rate

2.11.1.7 Balance Alarm

This function is used to monitor account balance.

 **NOTE**

The number of customer can be set by “System management > System parameter > SERVER_ALARM_CUSTOMER_BALANCE_MAX_SIZE”.

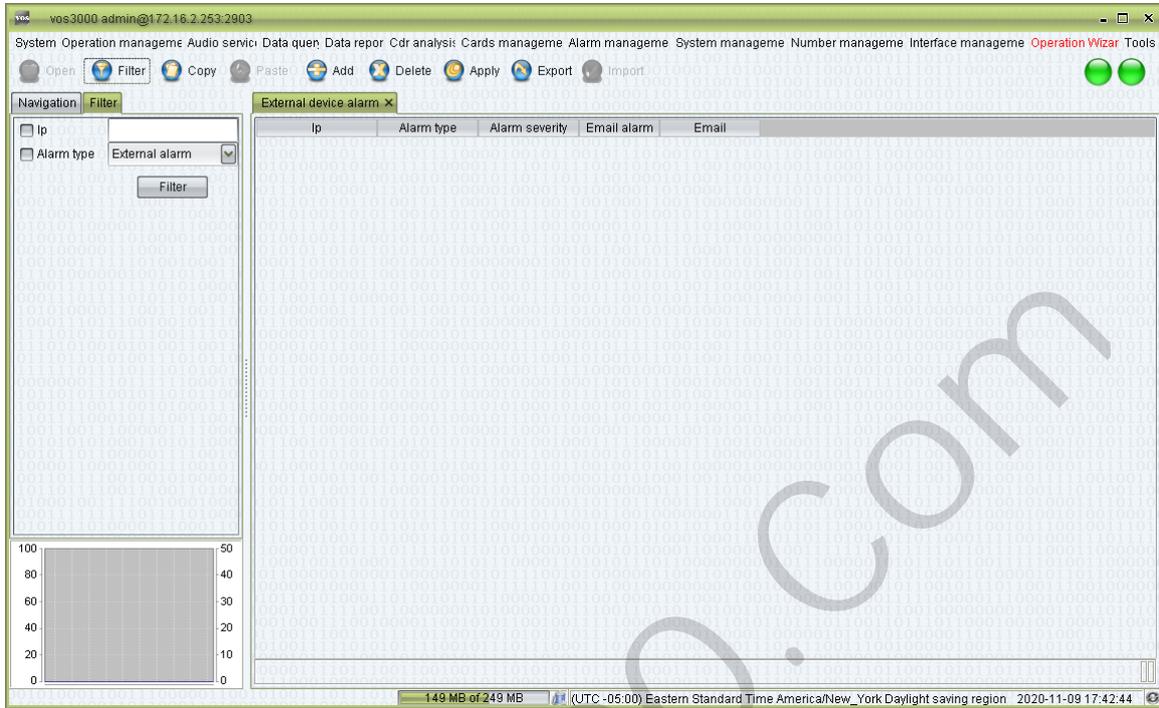


How to Start

- Double-click “Navigation > Alarm management > Alarm settings > Balance alarm”

2.11.1.8 External Device Alarm

This function is used to monitor external device.

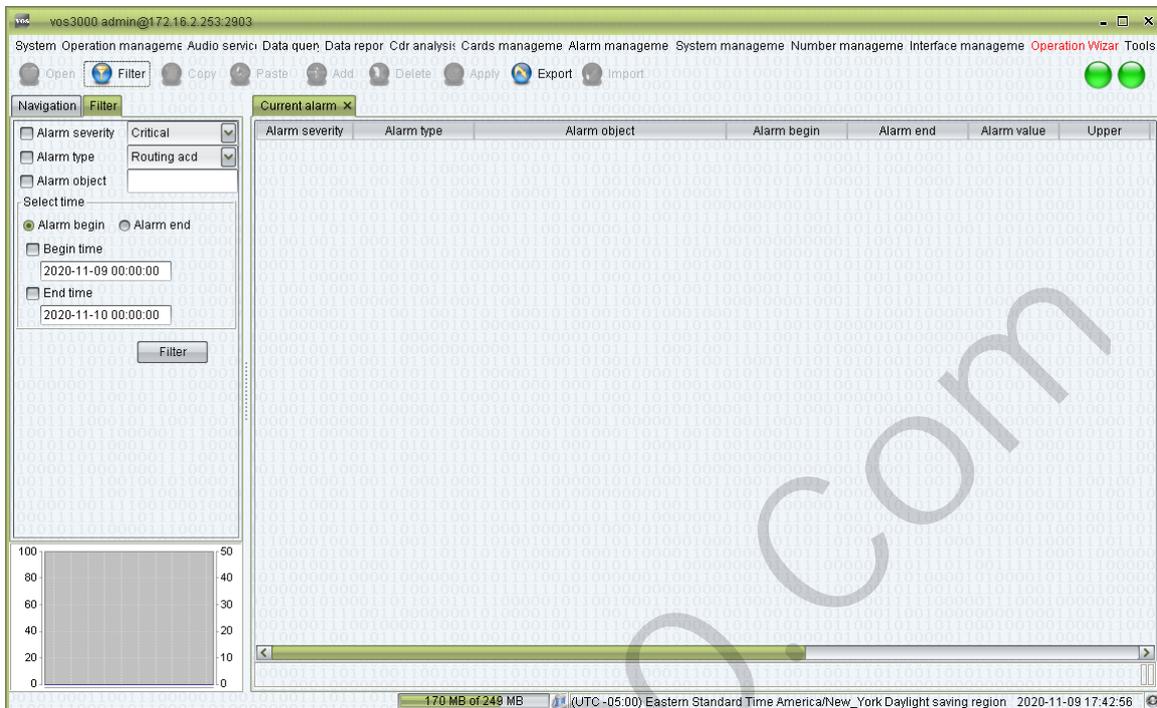


How to Start

- Double-click “Navigation > Alarm management > Alarm settings > External device alarm”

2.11.2 Current Alarm

This function is used to manage current alarm.



How to Start

- Double-click “Navigation > Alarm management > Current alarm”

Table Items

- Alarm severity
- Alarm type
- Alarm object
- Alarm begin
- Alarm end
- Alarm value
- Upper
- Lower
- Ack user
- Ack time
- Information
- Memo

NOTE

Begin time of the current alarm records the time when the alarm occurred for the first time. If the alarm object reports the alarm again, the start time does not change, and the alarm value uses the latest reported alarm value. After the alarm ends, if an alarm of the same alarm object occurs again, the ended alarm will be automatically cleared and placed in the historical alarm. The new alarm will be the current alarm.

Some current alarms may not have the alarm end time forever, and need to be cleared manually, such as "illegal call" "standby enable" "database" "call duration" "login limited" "financial loss",etc.

Right-Click Menu

- Confirm: input memo to confirm alarm.
- Empty: clear current alarm, and then change the alarm into history.

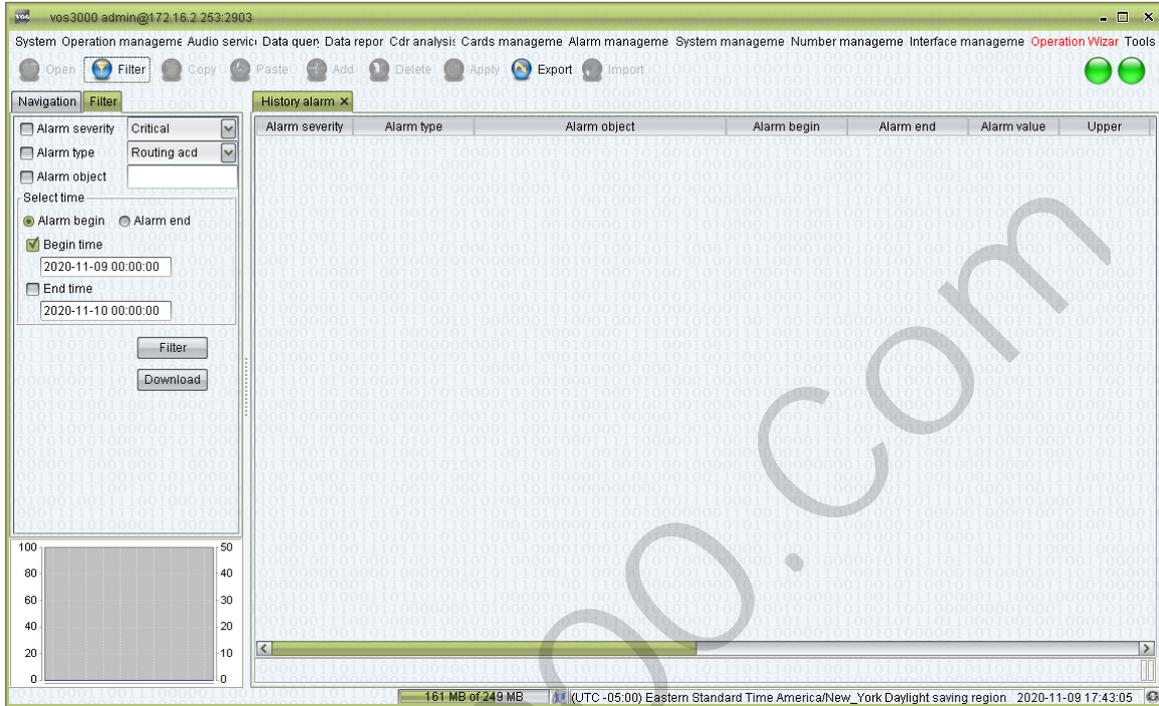
VOS3000.COM

2.11.3 History Alarm

This function is used to query history alarm.

 **NOTE**

If the current alarm, which is stopped, occurs again, the former alarm will be become history alarm.



How to Start

- Double-click “Navigation > Alarm management > History alarm”

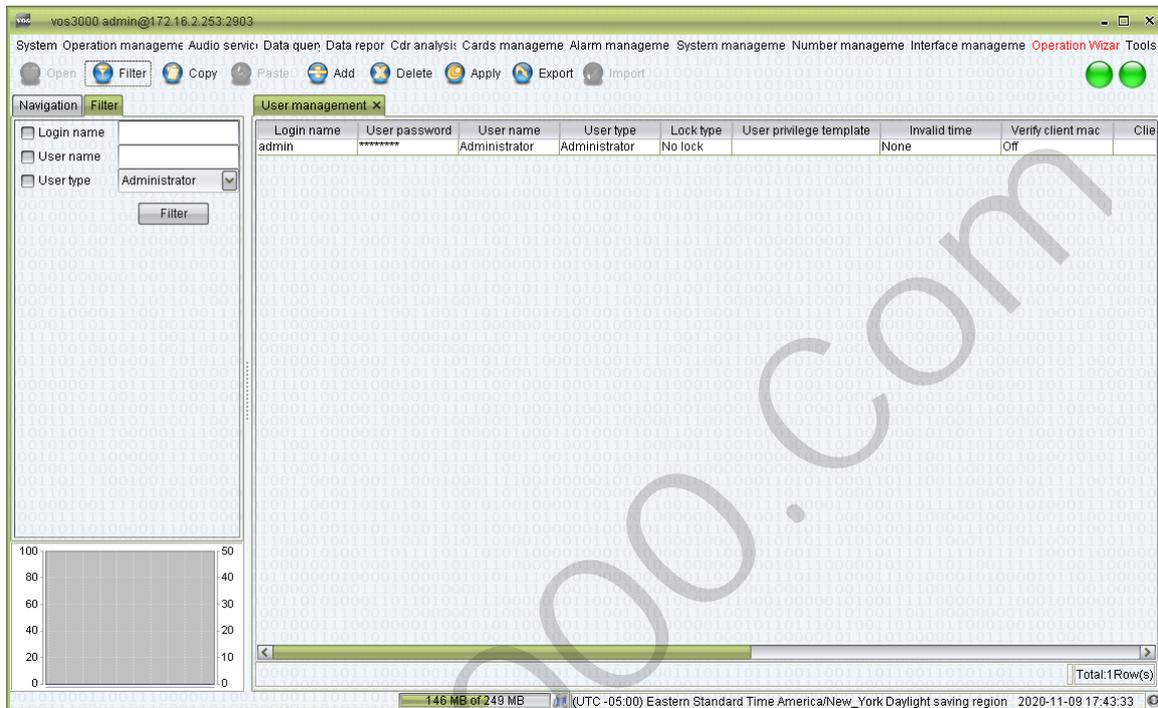
Table Items

- Alarm severity
- Alarm type
- Alarm object
- Alarm begin
- Alarm end
- Alarm value
- Upper
- Lower
- Ack user
- Ack time
- Memo
- Cleared user
- Cleared time

2.12 System Management

2.12.1 User Management

This function is used to manage user.



How to Start

- Double-click “Navigation > System management > User management”

Table Items

- Login name: the id used for login.
- User password: the login password.
- User name: the name of the user.
- User type:
 - Administrator: users with all authorizations.
 - Operator: users with certain authorizations for operations.
 - Agent: users that are only allow viewing the accounts, rate and service packages.
- Lock type:
 - No lock: user permissions can be used normally.
 - Locked: user permission cannot be used.
- Authorization: see below.
- Invalld time: after this time the user is not allowed to log in, the login user is often used for temporary.
- Verify client mac:

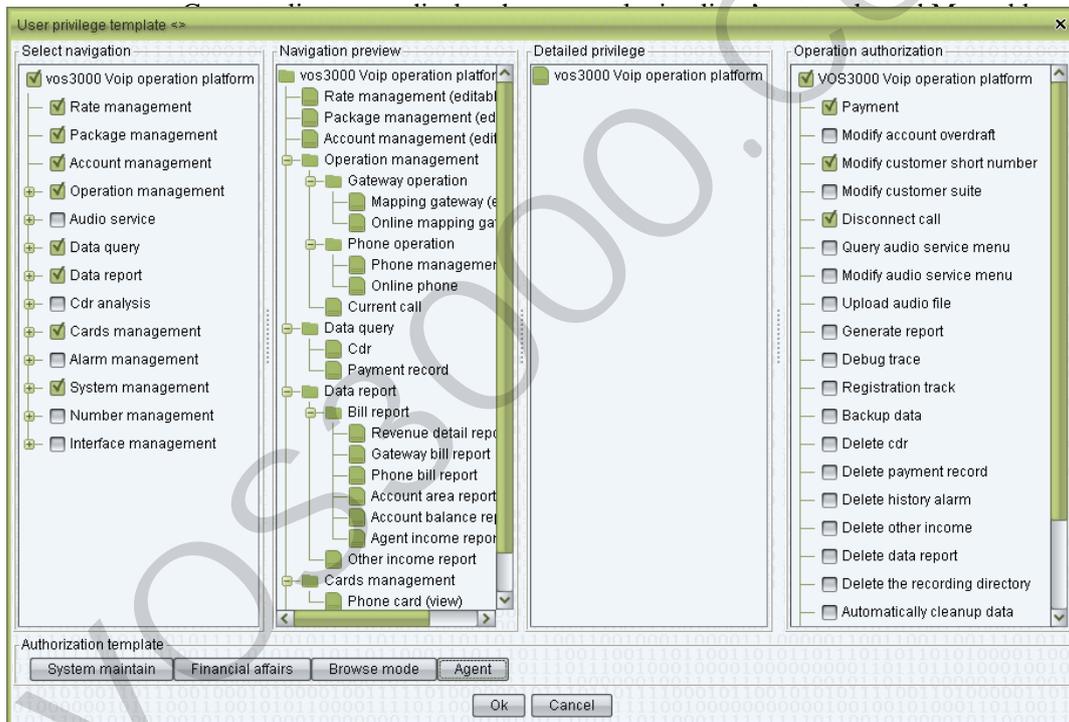
- Off: Does not verify client network card Mac address.
- Trigger alarm: if someone login vos with a network card Mac address which is not in <Client Mac List>,an alarming will be trigger when configured <Login Mac Restricted> warning in <System Alarm>.
- Ban login: forbid user to login vos if the login client has a network card Mac address which is not in <Client Mac List>.
- Client mac list: fill in network card Mac address.
- Memo: comments about the user.
- Dynamic password: choose user and right click to enable or close this feature.



NOTE

There will be an “Extraction code” window after you enable this feature, bind “app” with “Extraction code”, then you can use dynamic password to login vos client.

- App download link:
 - http://www.VOS3000.Com/chs/support/downloads/vos_otp.apk
 - Last login: last login time.
 - Last change password: last change password time.



- Select navigation: define user’s navigation tree.
- Navigation preview: preview user’s navigation tree, double click to change privilege (view or editable).
- Detailed privilege: display the detailed content of navigation tree, double click to change privilege (view or editable).
- Operation authorization: define the allowed operation privilege by user.



NOTE

Users can specify interfaces and operations available for a non-administrator user. All settings come into effect immediately.

- Authorization template: several templates for authorizations are provided. Users can select a template and then tune the configurations.

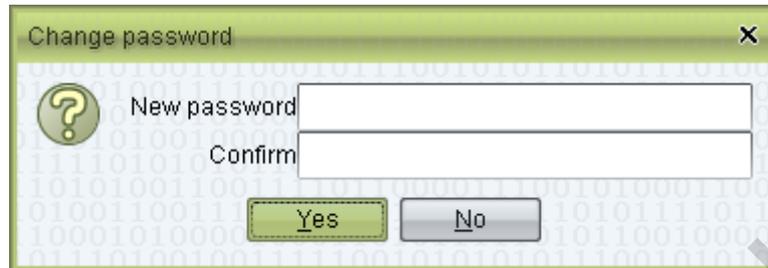


NOTE

Users of the “agent” type who have the authorization to create users will be able to see all the users they created in the table, while other users are invisible to them. It is the same when they specify the availability of rates, packages, and accounts to other users.

Right-Click Menu

- Change password

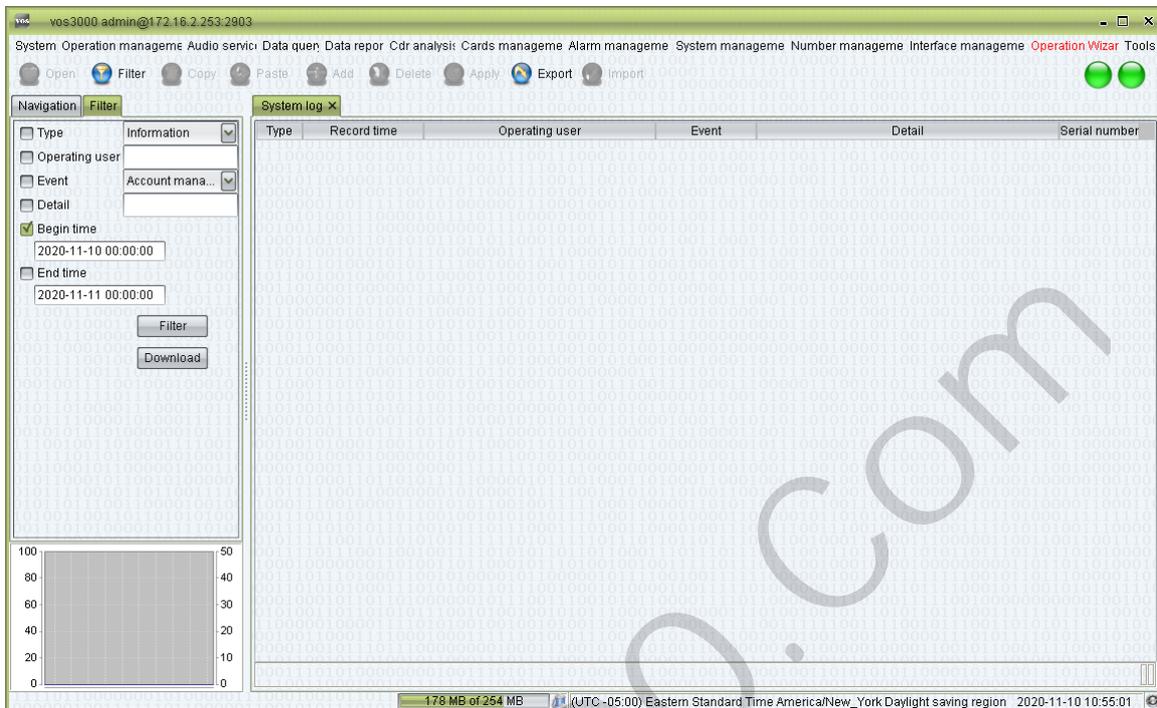


Password requirements:

- at least 6 characters (at least 8 characters for administrator)
- at least 2 of the following character combination:
 - at least one lowercase letter
 - at least one uppercase letter
 - at least one number
 - at least on special character: `~!@#\$\$%^&*()-_+=\|[{ }];:","<.>/? and space (space cannot be used as starting or ending character.)
- cannot be the same as the name or the reverse of name

2.12.2 System Log

This function is used to query system log.



How to Start

- Double-click “Navigation > System management > System log”

Table Items

- Type: Information/General/Error
- Record time: log time.
- Operating User
- Event
- Detail
- Serial number

Right Click Menu

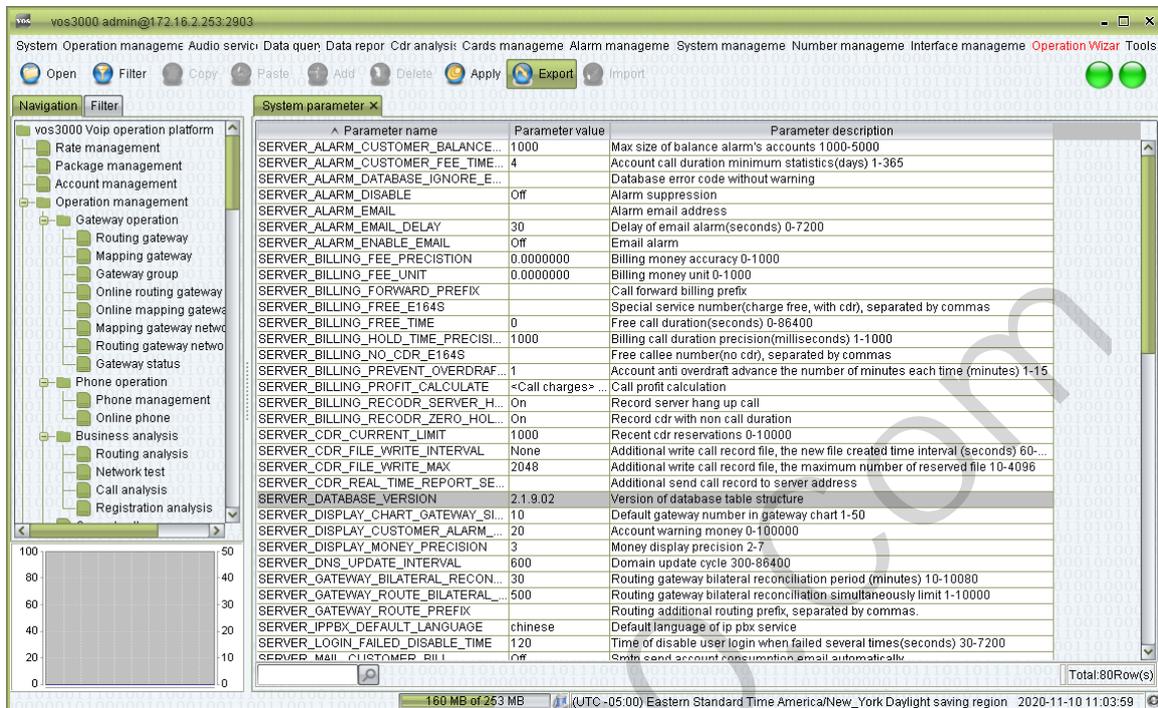
- Detail

This function is used to show operating user and operation’s detail.



2.12.3 System Parameter

This function is used to configure parameters.



How to Start

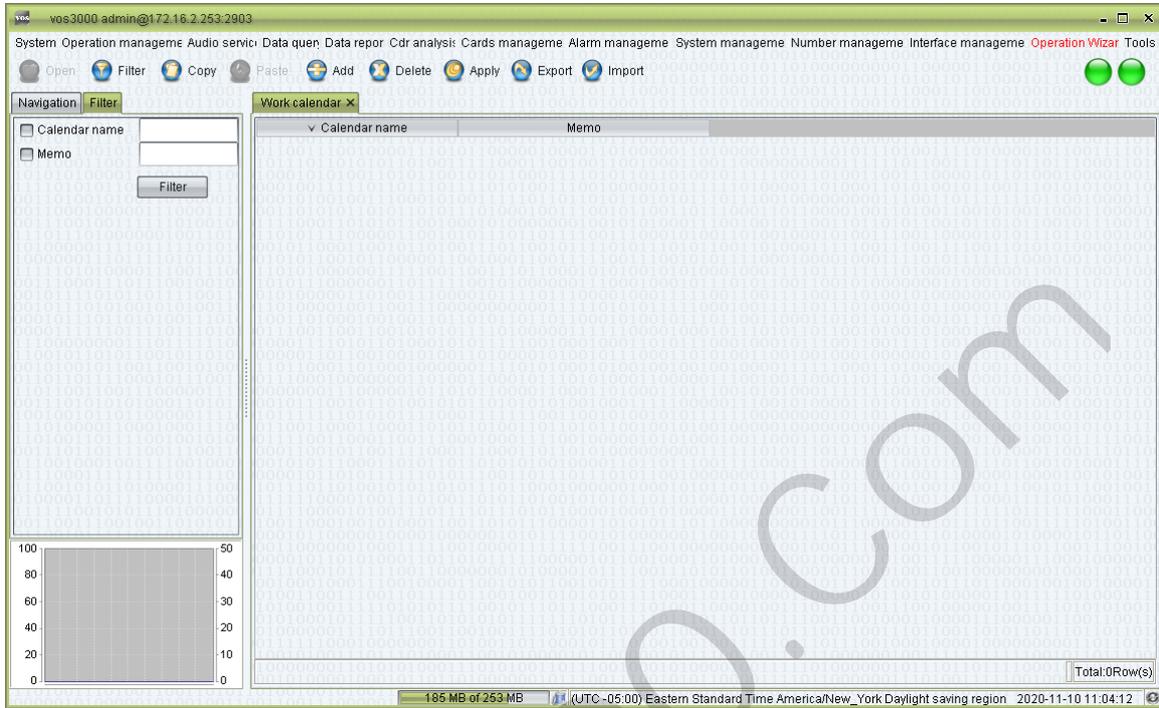
- Double-click “Navigation > System management > System parameter”

Table Items

- Parameter name
- Parameter value
- Parameter description

2.12.4 Work Calendar

This function is used to define working and non-working hours.



How to Start

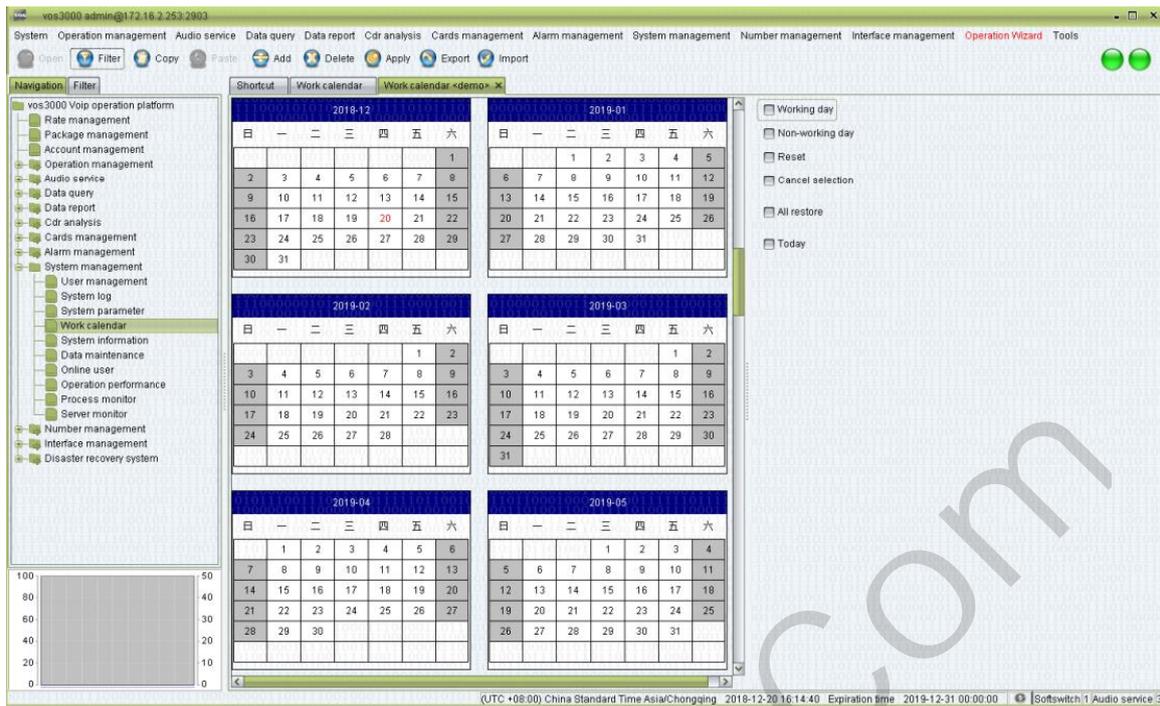
Double-click “Navigation > System management > Work calendar”

Table Items

- Calendar name
- Memo

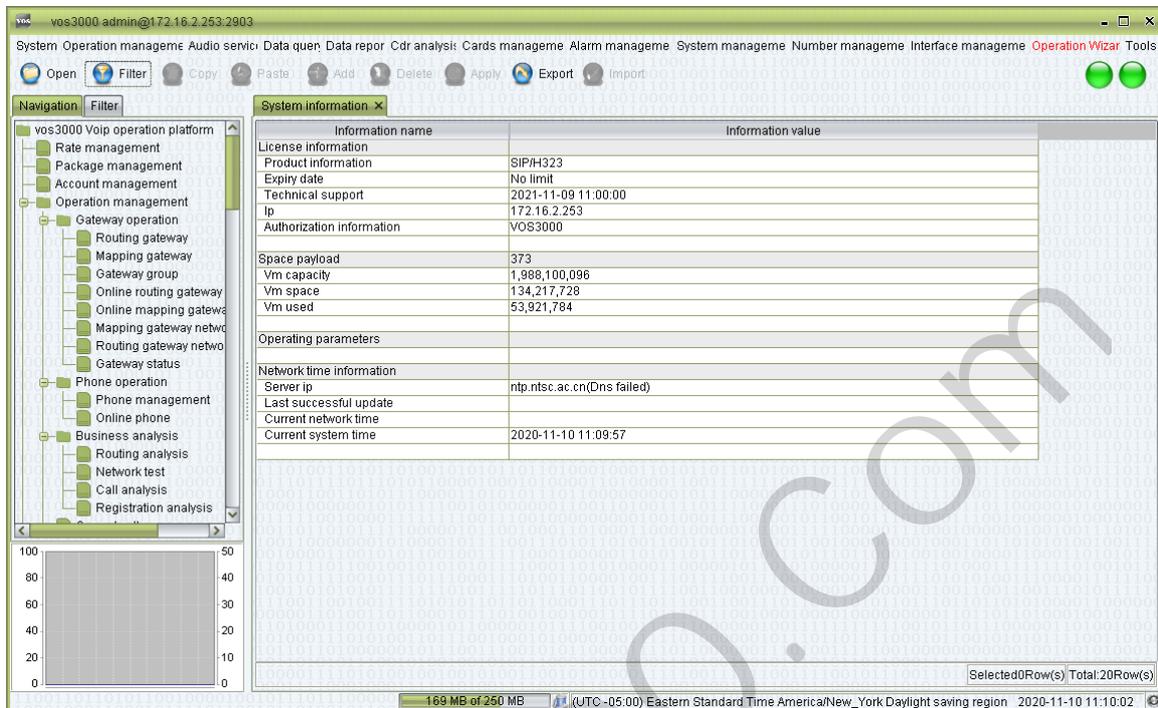
Work Calendar

This function is used to match the use of the account "Suppressing all duration too long".



2.12.5 System Information

This function is used to show server's information.

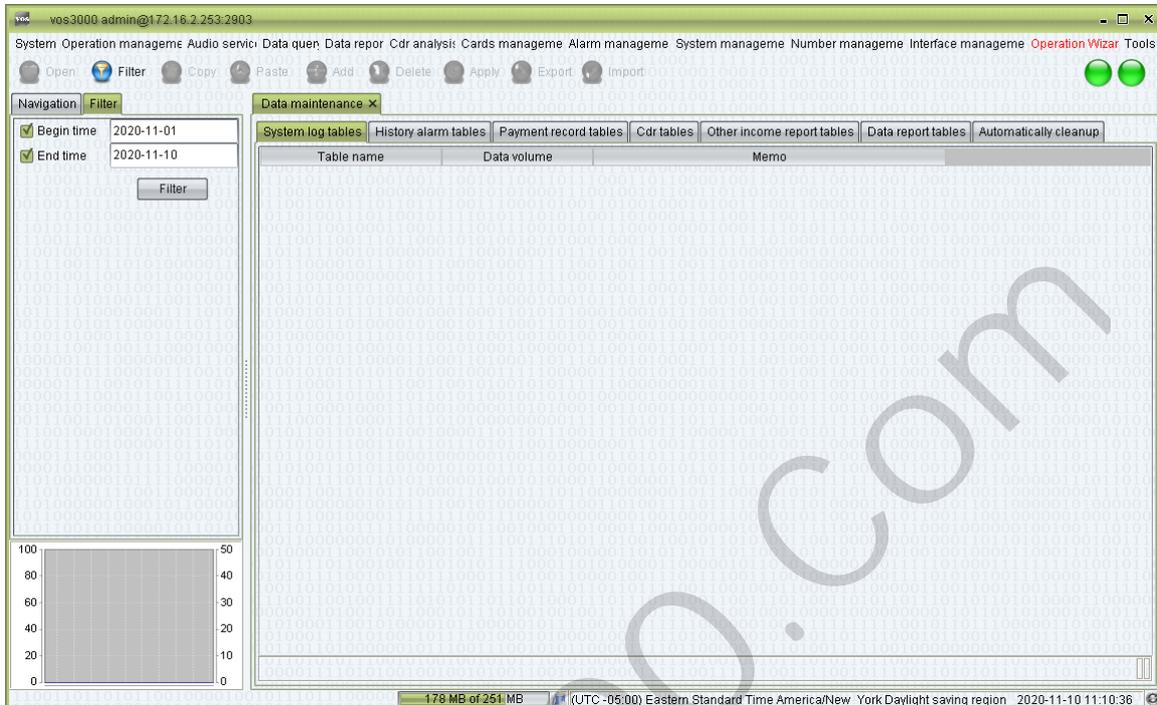


How to Start

- Double-click “Navigation > System management > System information”

2.12.6 Data Maintenance

2.12.6.1 System Log Tables



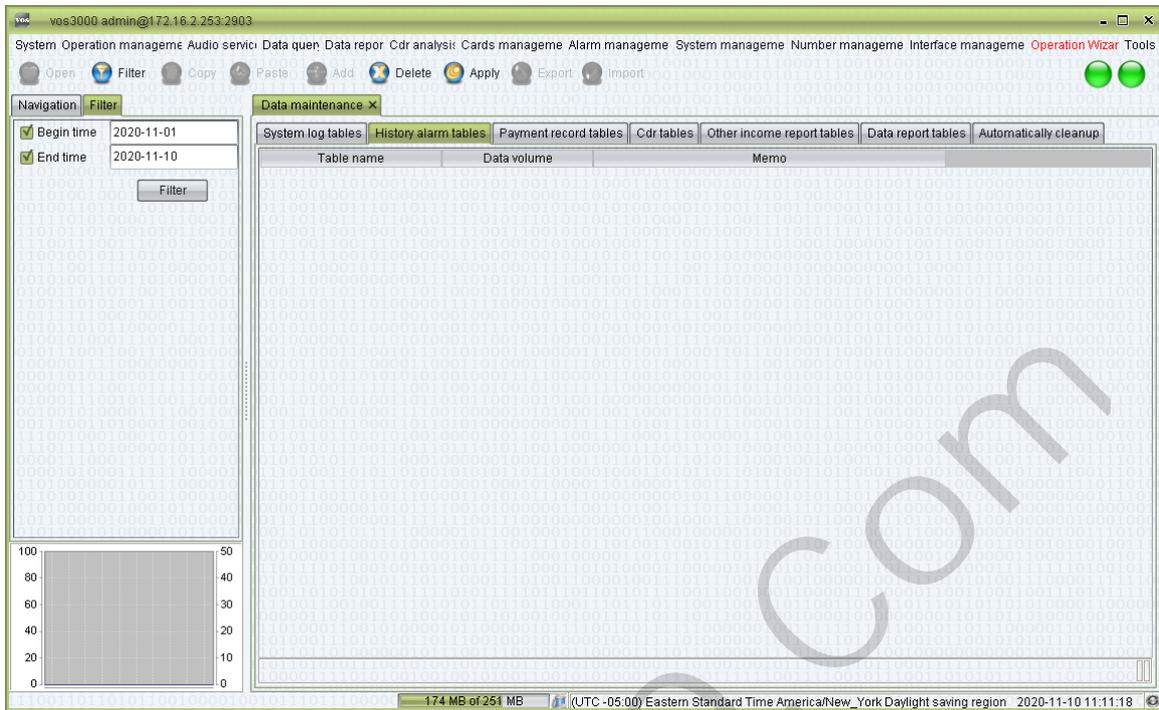
How to Start

- Double-click “Navigation > System management > Data maintenance”

Table Items

- Table name: suffix is the date of system log.
- Data volume: number of records.
- Memo

2.12.6.2 History Alarm Tables



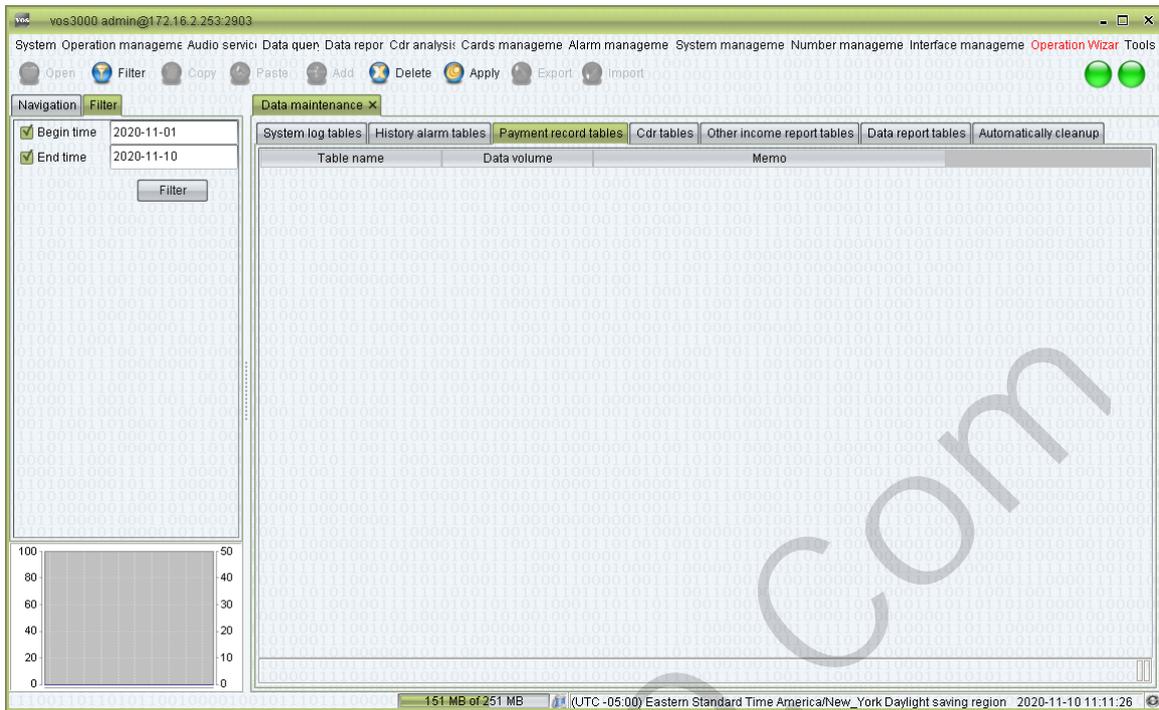
How to Start

- Double-click “Navigation > System management > Data maintenance”

Table Items

- Table name: suffix is the date of history alarm.
- Data volume: number of records.
- Memo

2.12.6.3 Payment Record Tables



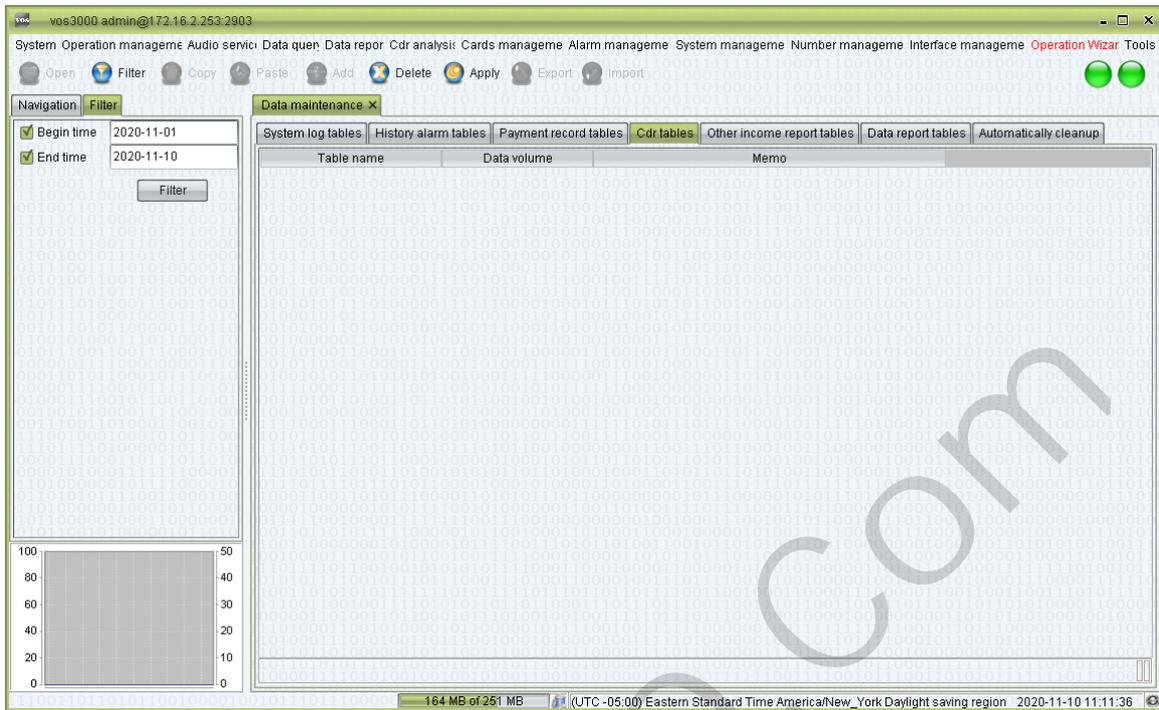
How to Start

- Double-click “Navigation > System management > Data maintenance”

Table Items

- Table name: suffix is the date of payment record.
- Data volume: number of records.
- Memo

2.12.6.4 CDR Tables



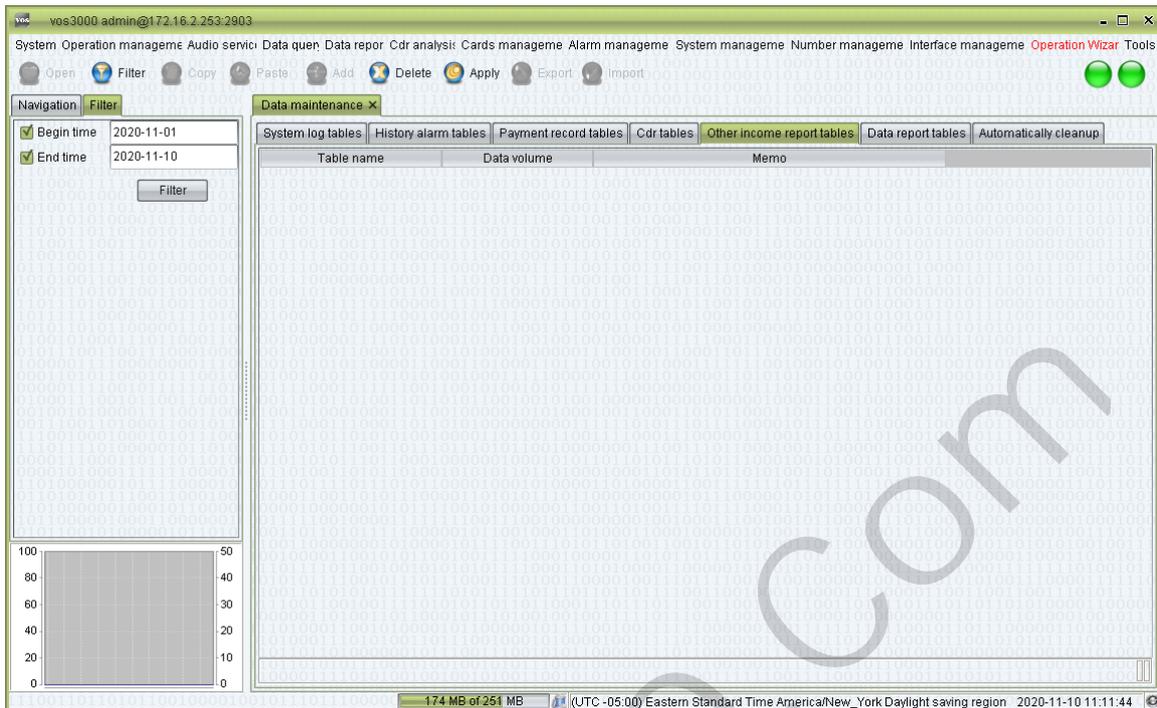
How to Start

- Double-click “Navigation > System management > Data maintenance”

Table Items

- Table name: suffix is the date of CDR.
- Data volume: number of records.
- Memo

2.12.6.5 Other Income Report Tables



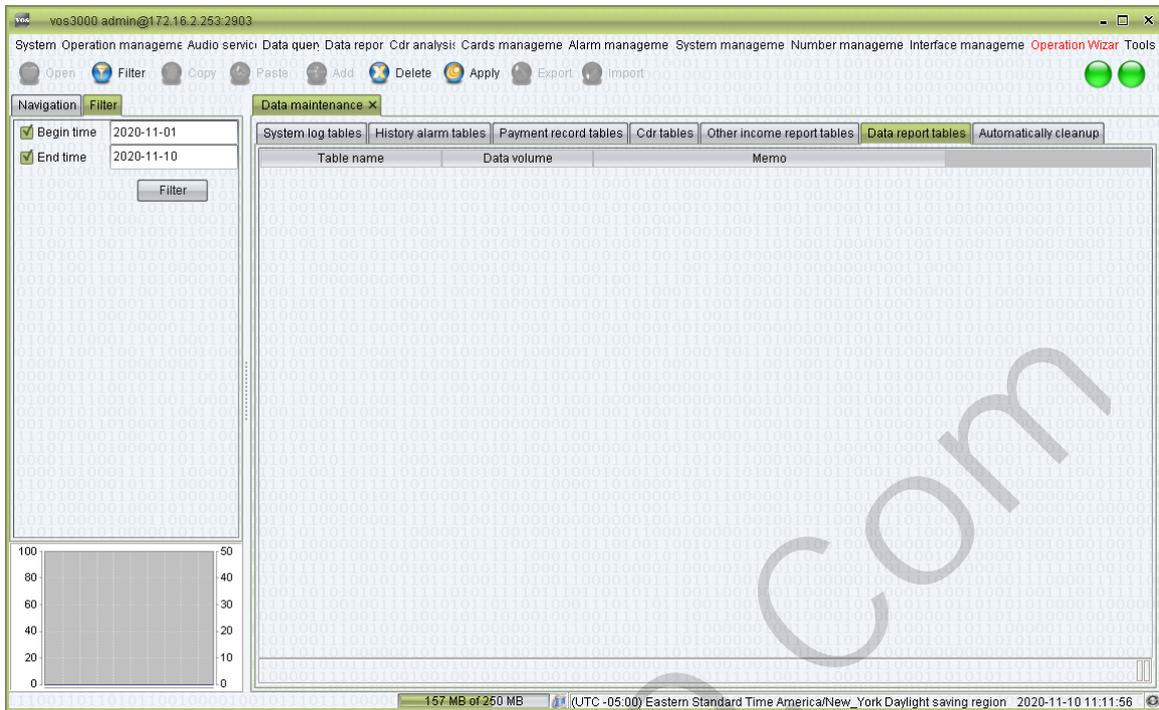
How to Start

- Double-click “Navigation > System management > Data maintenance”

Usage

- Use “Filter” to get current settings.
- If auto cleanup is enabled, vos will cleanup out of date data every day, including account/account’s gateway/account’s phone.

2.12.6.6 Data Report Tables



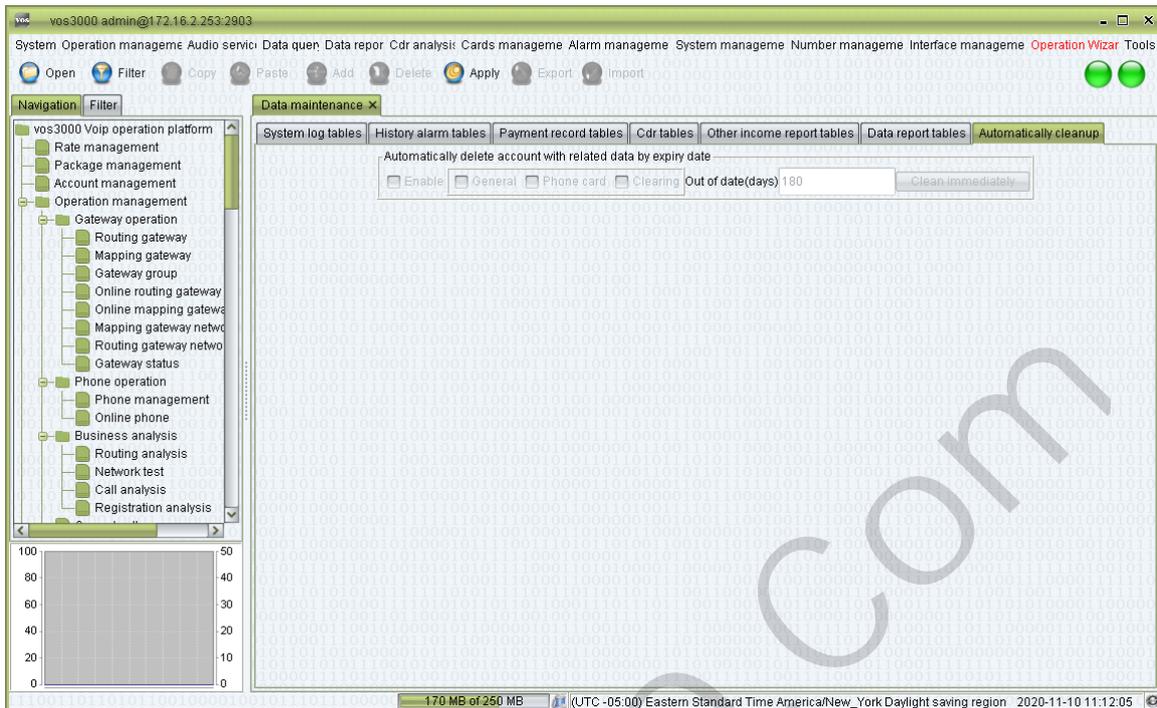
How to Start

- Double-click “Navigation > System management > Data maintenance”

Table Items

- Table name: suffix is the date of report.
- Data volume: number of records.
- Memo

2.12.6.7 Automatically Cleanup



How to Start

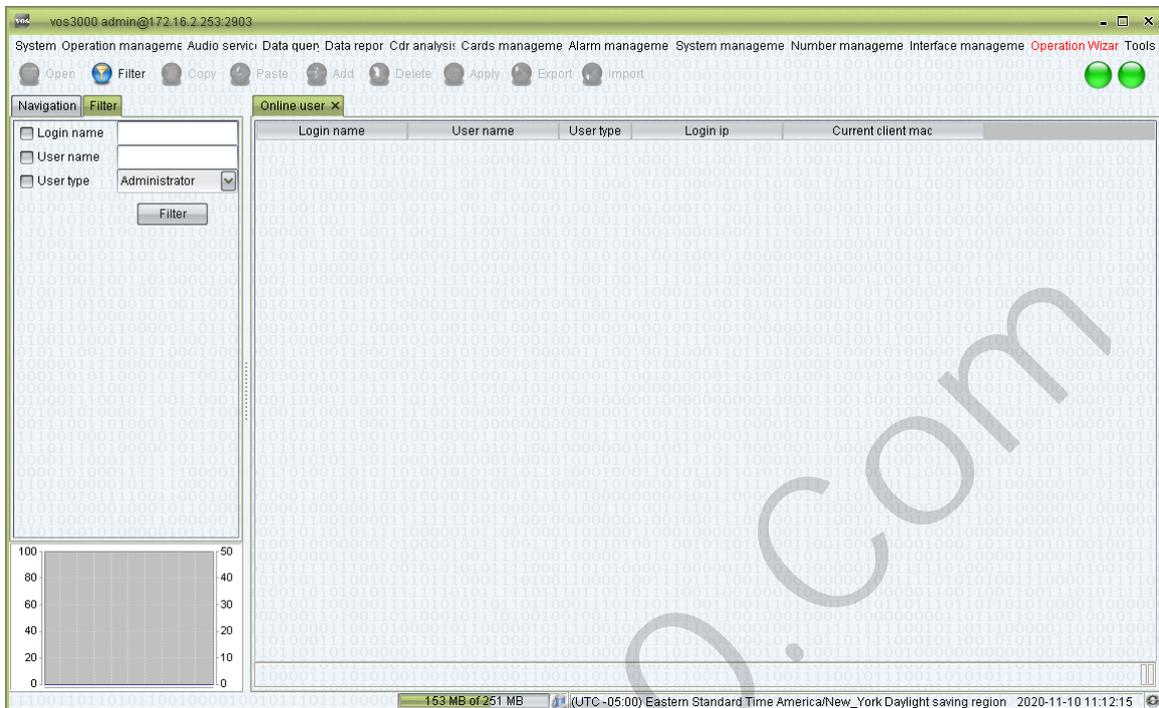
- Double-click “Navigation > System management > Data maintenance”

Usage

- Use “Filter” to get current settings.
- If auto cleanup is enabled, vos will cleanup out of date data every day, including account/account’s gateway/account’s phone.

2.12.7 Online User

This function is used to query online user.

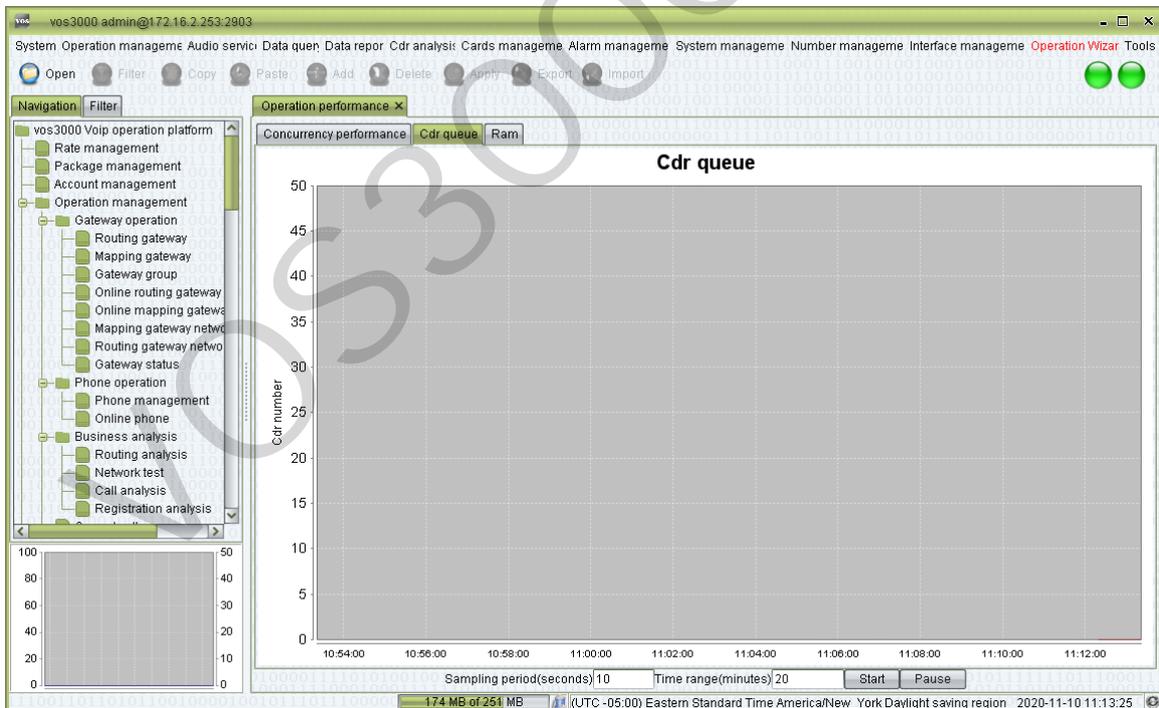
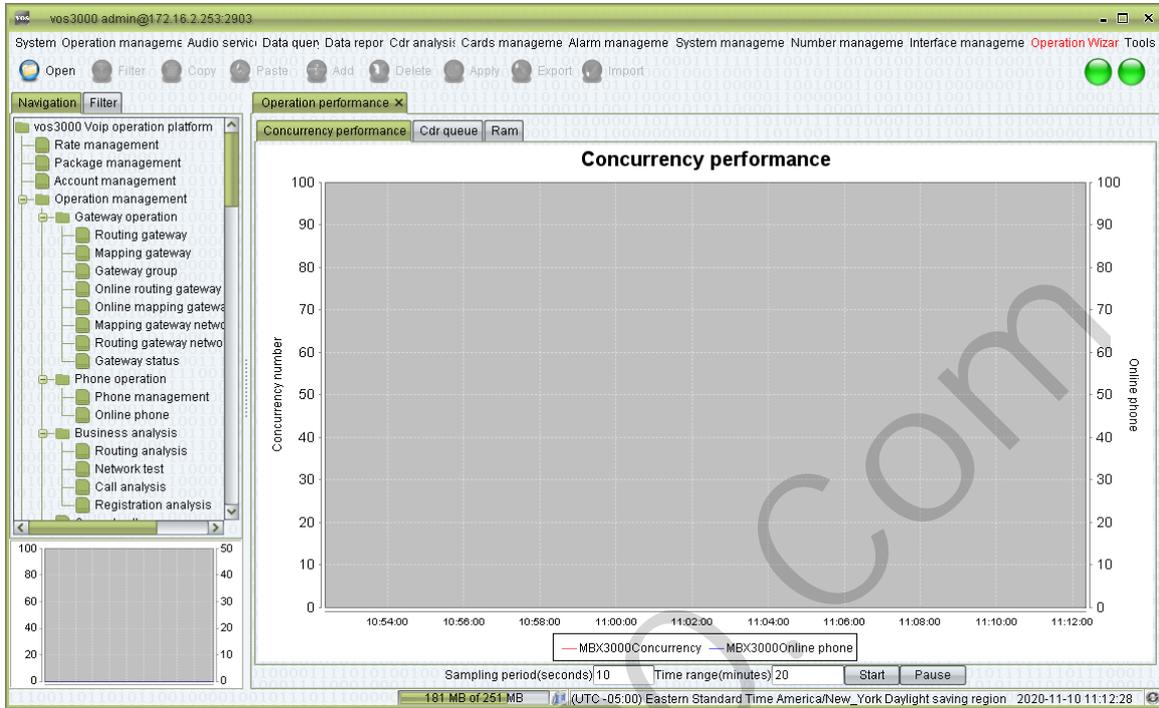


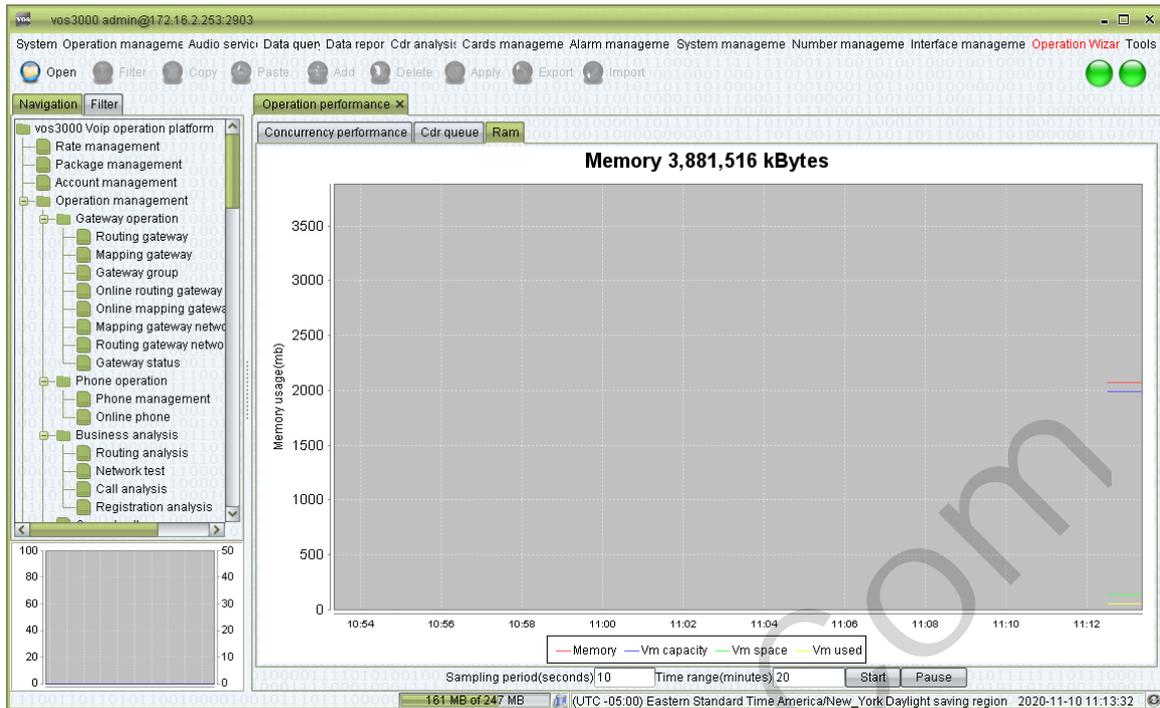
How to Start

- Double-click "Navigation > System management > Online user"

2.12.8 Operation Performance

This function is used to monitor system performance.



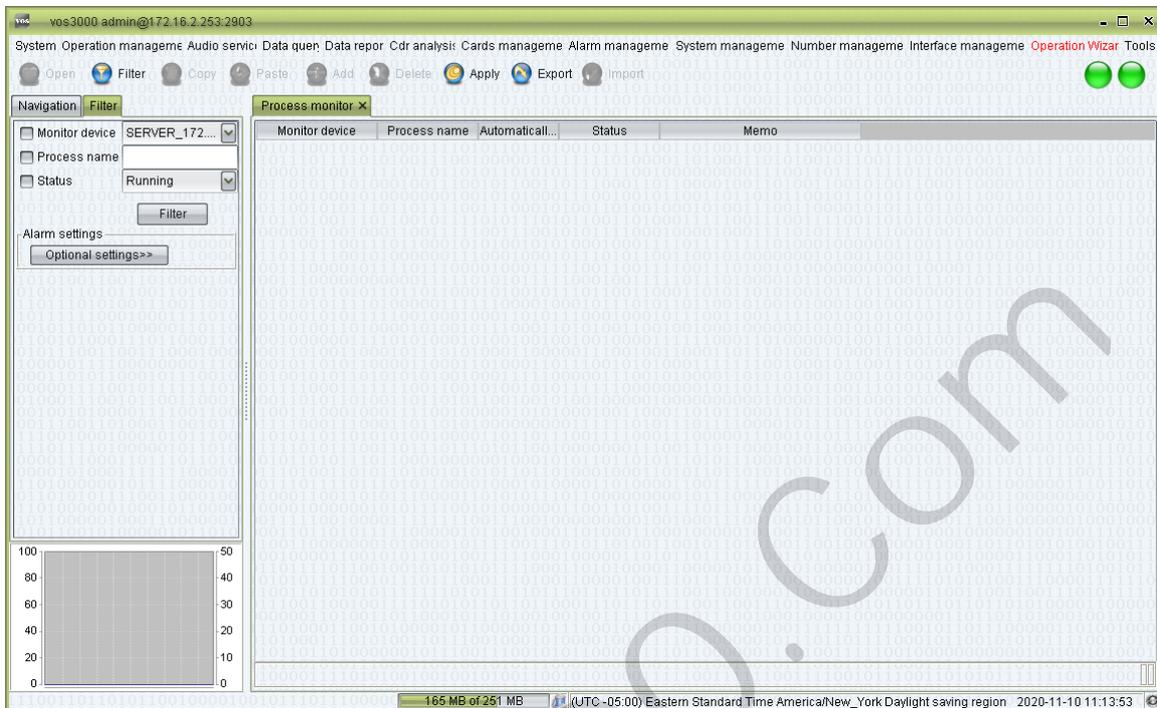


How to Start

- Double-click “Navigation > System management > Operation performance”

2.12.9 Process Monitor

This function is used to monitor process.



How to Start

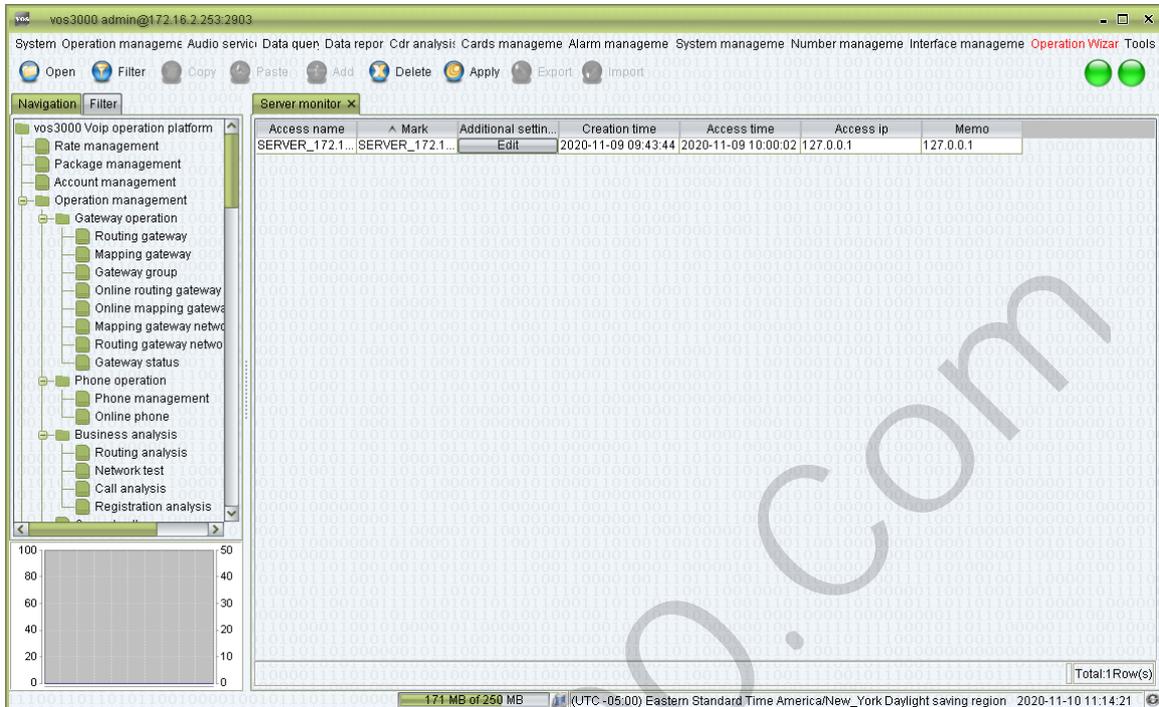
Double-click “Navigation > System management > Process monitor”

Usage

- Monitor device
- Process name
- Automatically restart
 - On: When process stop(not stoped by manual),system will restart this process
 - Off: When process stop,system don't do anything
- Status
- Memo

2.12.10 Server Monitor

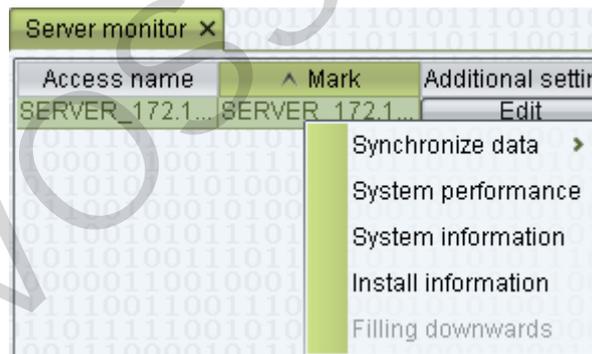
This function is used to monitor server.



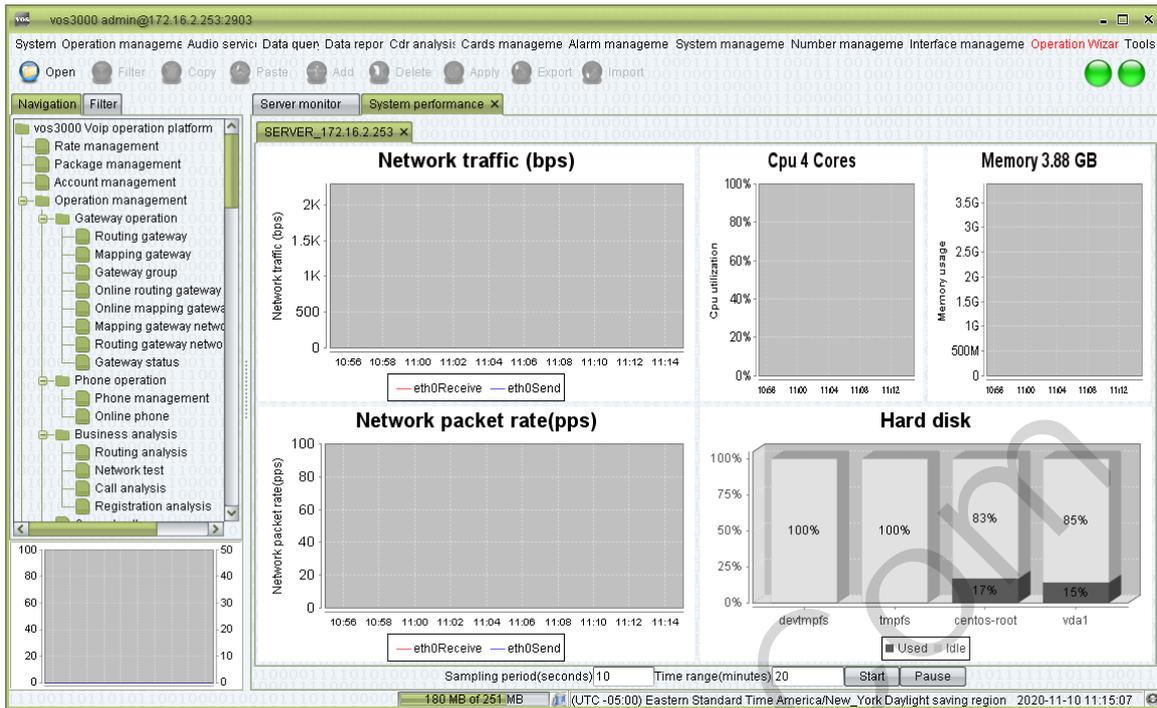
How to Start

Double-click “Navigation > System management > Server monitor”

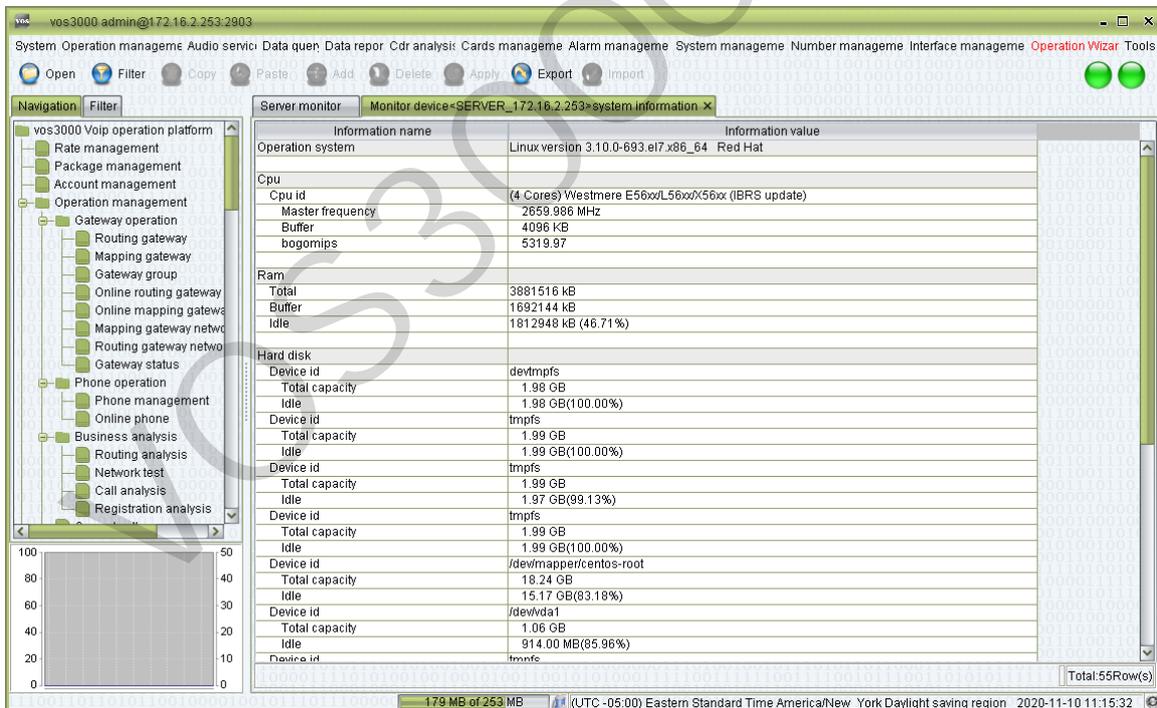
Right click menu



- Synchronize data
- System performance: current status about cpu,network,memory and hard disk



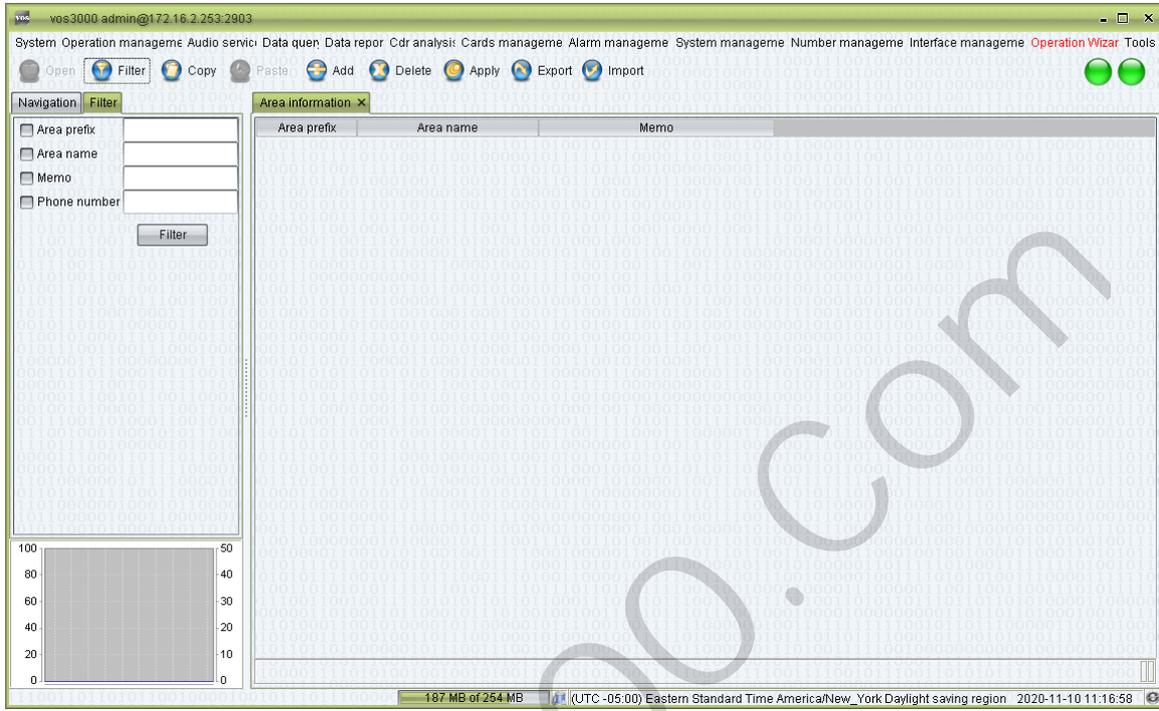
- System information: cpu, network, hard disk, ram, time, os and process details information



2.13.2 Area Information

This function is used to define area prefix's area name, country code, dialing prefix.

This setting will be used in rate management to show prefix's area name, using longest match.



How to Start

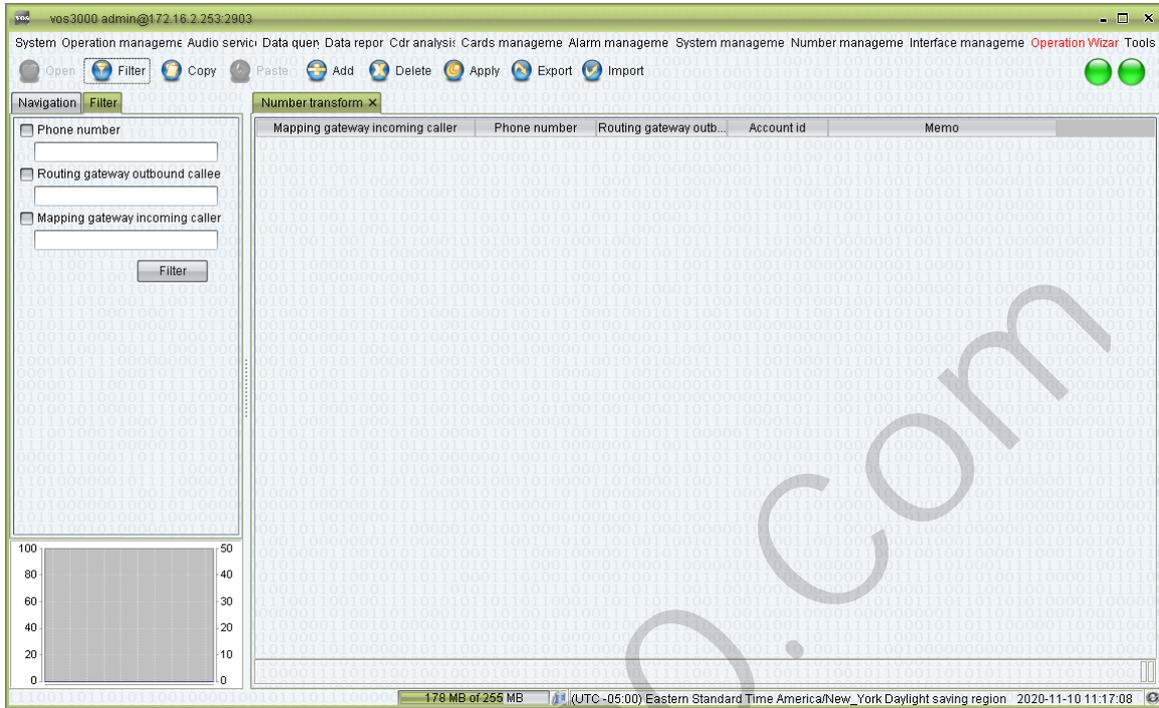
- Double-click “Navigation > Number management > Area information”

Table Items

- Area prefix
- Area name
- Memo

2.13.3 Number Transform

This function is used to manage number transform.



How to Start

- Double-click “Navigation > Number management > Number transform”

2.13.4 Black/White List Group

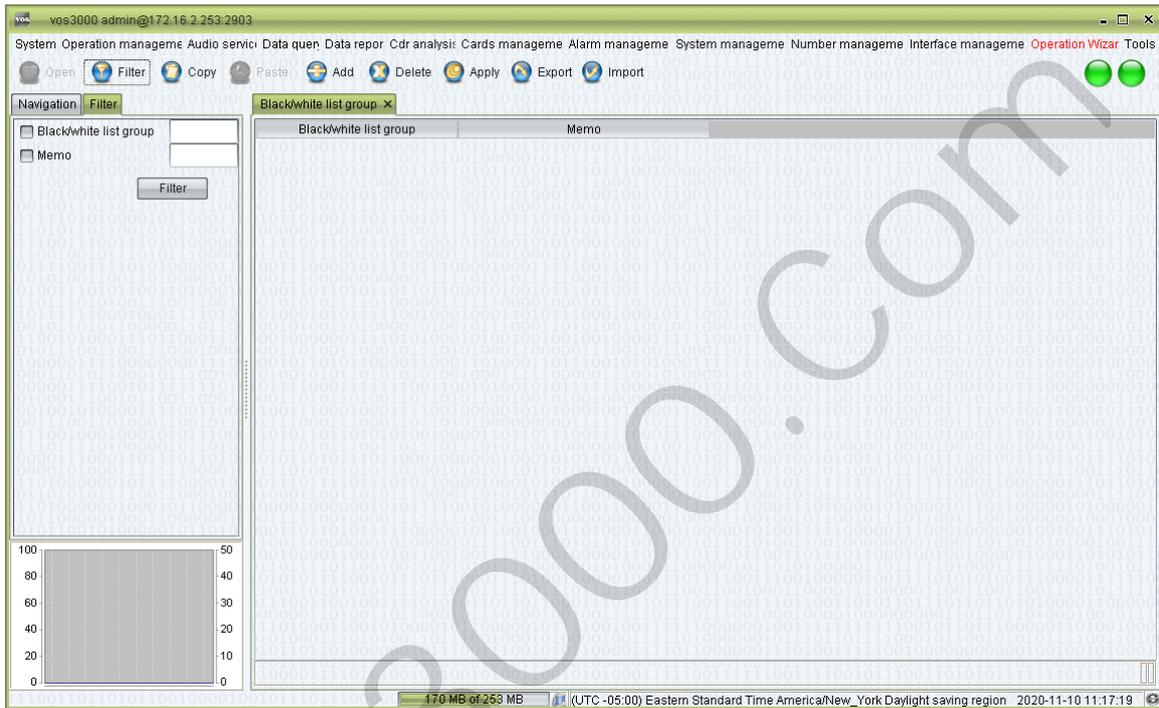
This function is used to define list group.

This setting will be used in caller/callee black/white list group of routing gateway, mapping gateway and phone.

 **TIP**

Black/white list group is full match, more efficient than prefix match.

For a large number match, use this function instead of prefix match.



How to Start

- Double-click “Navigation > Number management > Black/White List Group”

Table Items

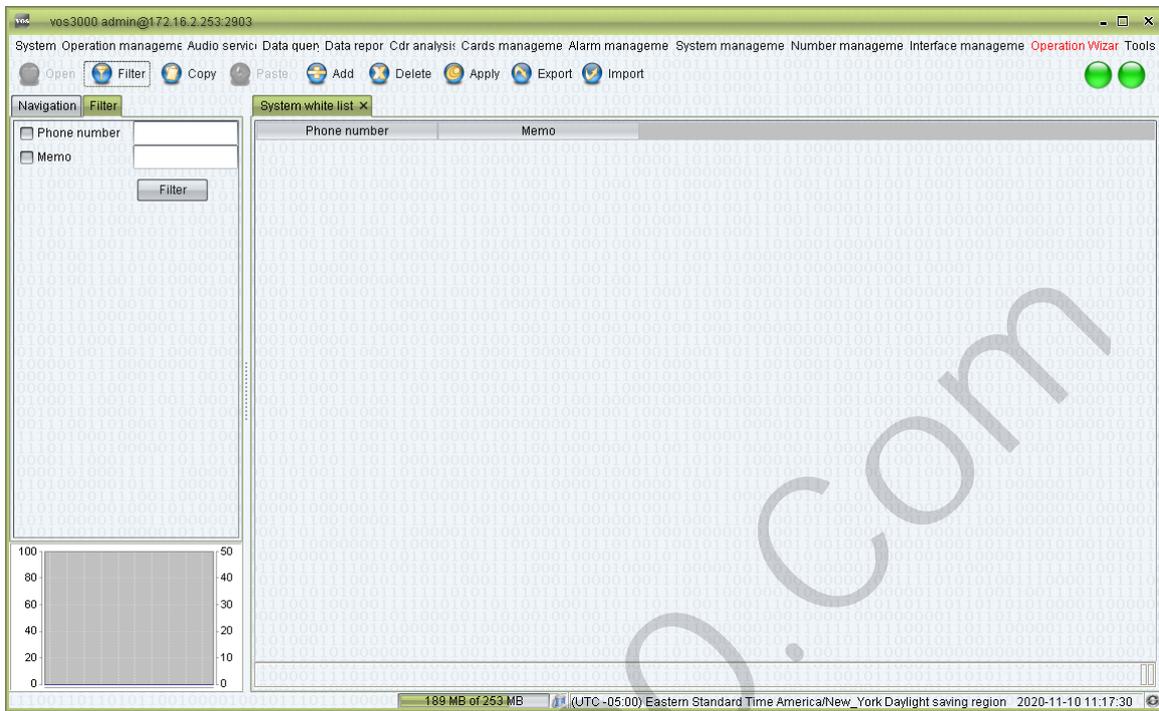
- Black/white list group: name of the group.
- Memo

Other Operation

- Double click black/white group to edit number list.

2.13.5 System White List

This function is used to manage system white list.



How to Start

- Double-click “Navigation > Number management > System white list”

Table Items

- Phone number
- Memo

Softswitch<MBX3000>settings

SIP Parameter H323 Parameter System parameter

Parameter name	Parameter value	Parameter de
SS_CALL_FORWARD_USING_ORIGINAL_CALLER	On	Use the original caller as f
SS_CALL_HOLD_KEY		Call hold key
SS_CALL_PICK_UP_KEY	*4	Call pickup key
SS_CALL_SERVICE_CANCEL_KEY	**	Call transfer cancel key
SS_CALL_TRANSFER_ASK_DISPLAY	Original caller	Ask call transfer display
SS_CALL_TRANSFER_ASK_KEY	*2	The inquisitorial call transf
SS_CALL_TRANSFER_END_KEY	#	Confirm button for call tran
SS_CALL_TRANSFER_NORMAL_DISPLAY	Original caller	Normal call transfer displa
SS_CALL_TRANSFER_NORMAL_KEY	*1	Start button for call transfer
SS_CALL_TRANSFER_REMOTE_RING_PASS_THRO...	On	Send color ring back tone v
SS_CALL_TRANSFER_WAIT_ACCESS_KEY	*3	The pickup call transfer sta
SS_CALL_TRANSFER_WAIT_ACCESS_TIMEOUT	120	Pickup call transfer timeou
SS_CDR_RECORD_ILLEGAL	On	Record illegal call
SS_DEFAULT_LOCAL_IP	Default	Default local address
SS_DTMF_TIMEOUT	5	Waiting time for automatic
SS_ENDPOINT_EXPIRE	3600	Terminal registration expir
SS_ENDPOINT_NAT_EXPIRE	120	Terminal registration expir
SS_ENDPOINT_REGISTER_REPLACE	On	Allow replace the current re
SS_GATEWAY_ASR_ROUTE_SORT_CONFIG	Before line usage	Position for routing gatewa
SS_GATEWAY_ASR_ROUTE_SORT_METHOD	First route	The value used by the asr
SS_GATEWAY_FEE_RATE_ROUTE_BEFORE_ASR	Off	Rate routing priority over as
SS_GATEWAY_FEE_RATE_ROUTE_SORT_CONFIG	Before line usage	Position for routing gatewa

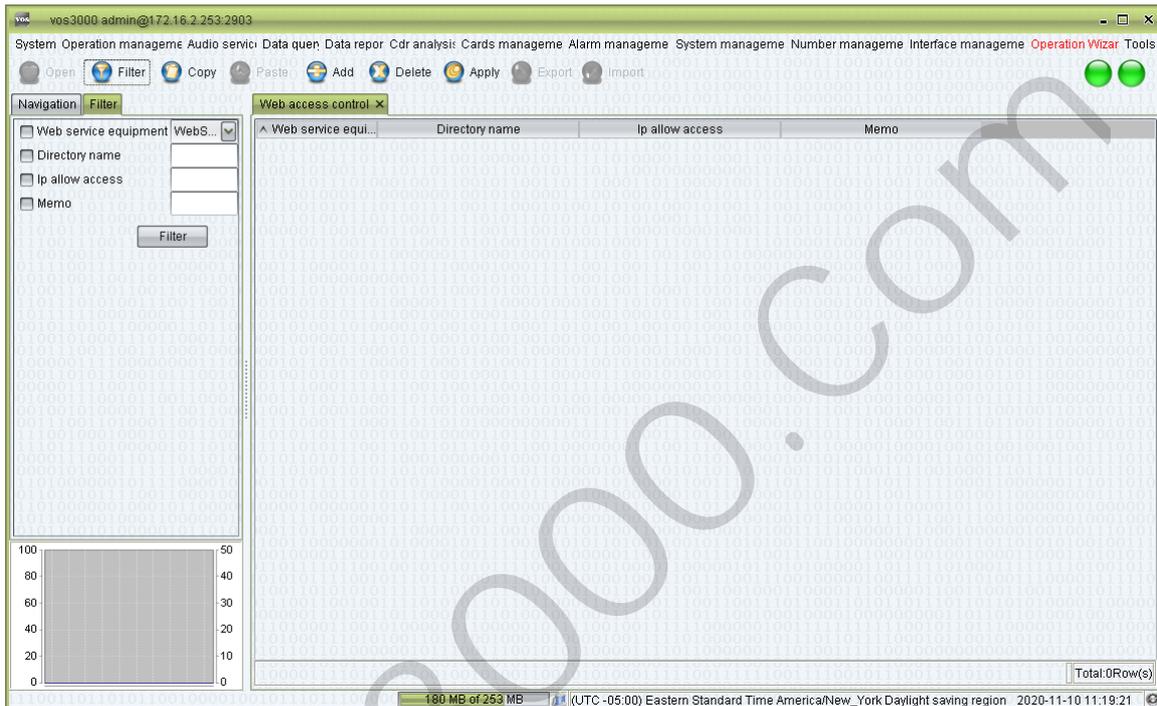
Selected 0 Row(s) Total: 88 Row(s)

Ok Cancel

2.14 Interface Management

2.14.1 Web Access Control

This function is used to manage external access ip.



How to Start

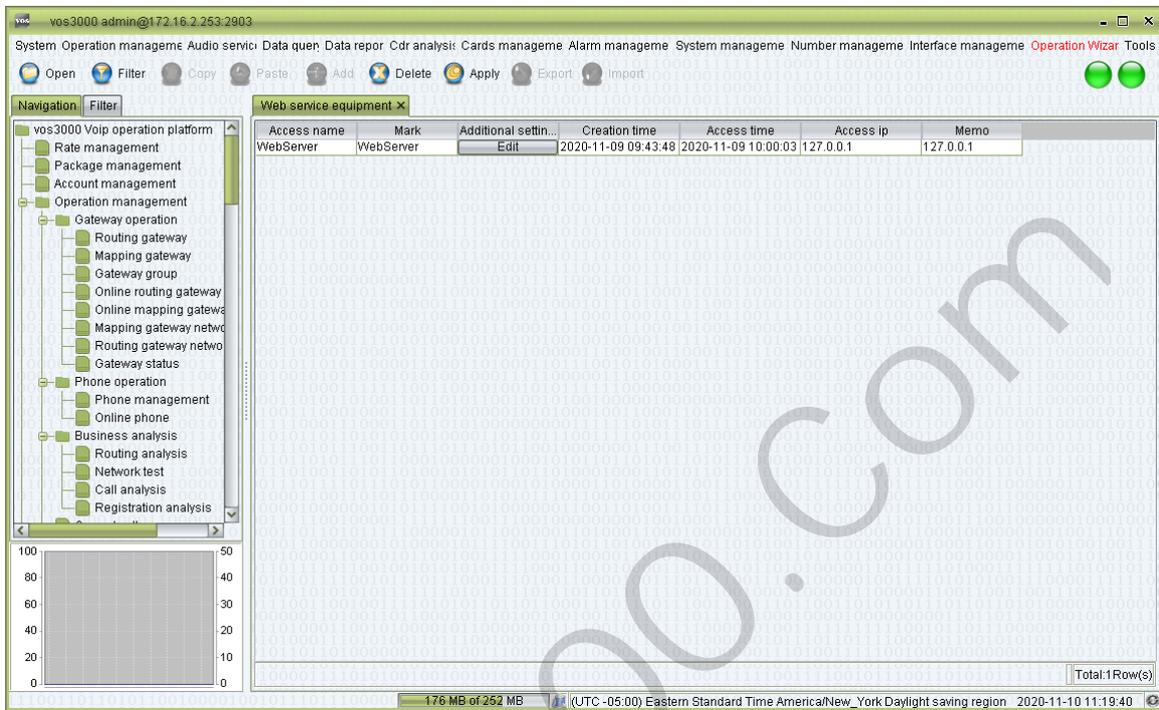
Double-click “Navigation > Interface management > Web access control”

Table Items

- Web service equipment
- Directory name
- Ip allow access
- Memo

2.14.2 Web Service Equipment

This function is used to query web service equipment.



How to Start

Double-click “Navigation > Interface management > Web service equipment”

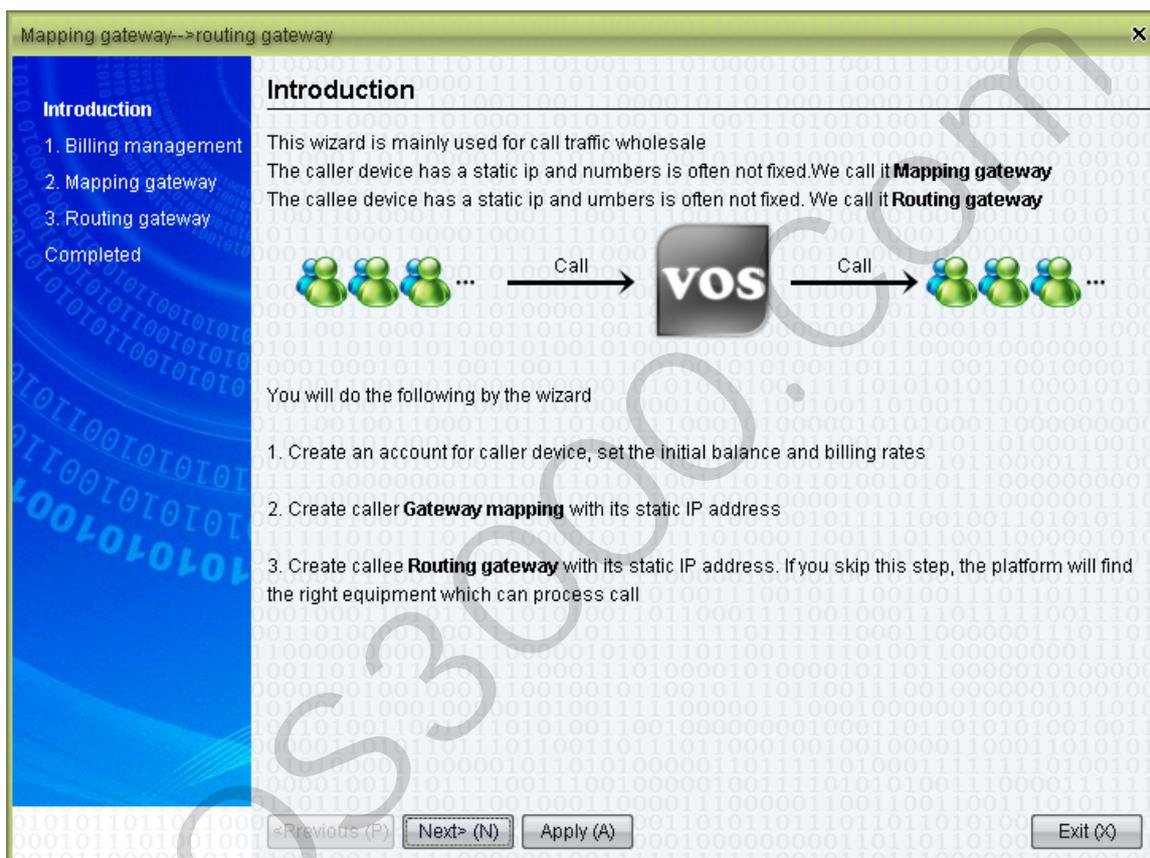
Table Items

- Access name
- Mark
- Additional setting
- Creation time
- Access time
- Access ip
- Memo

2.15 Operation Wizard

This function is used to quickly configure data.

Mapping gateway → routing gateway



Mapping gateway → phone

Mapping gateway-->phone

Introduction

1. Billing management
2. Mapping gateway
3. Callee phone
Completed

Introduction

This wizard is mainly used for outside numbers call platform number
The caller device has a static ip and numbers is often not fixed. We call it **Mapping gateway**
The callee use number and password register to platform through sip protocol. We call it **Phone**

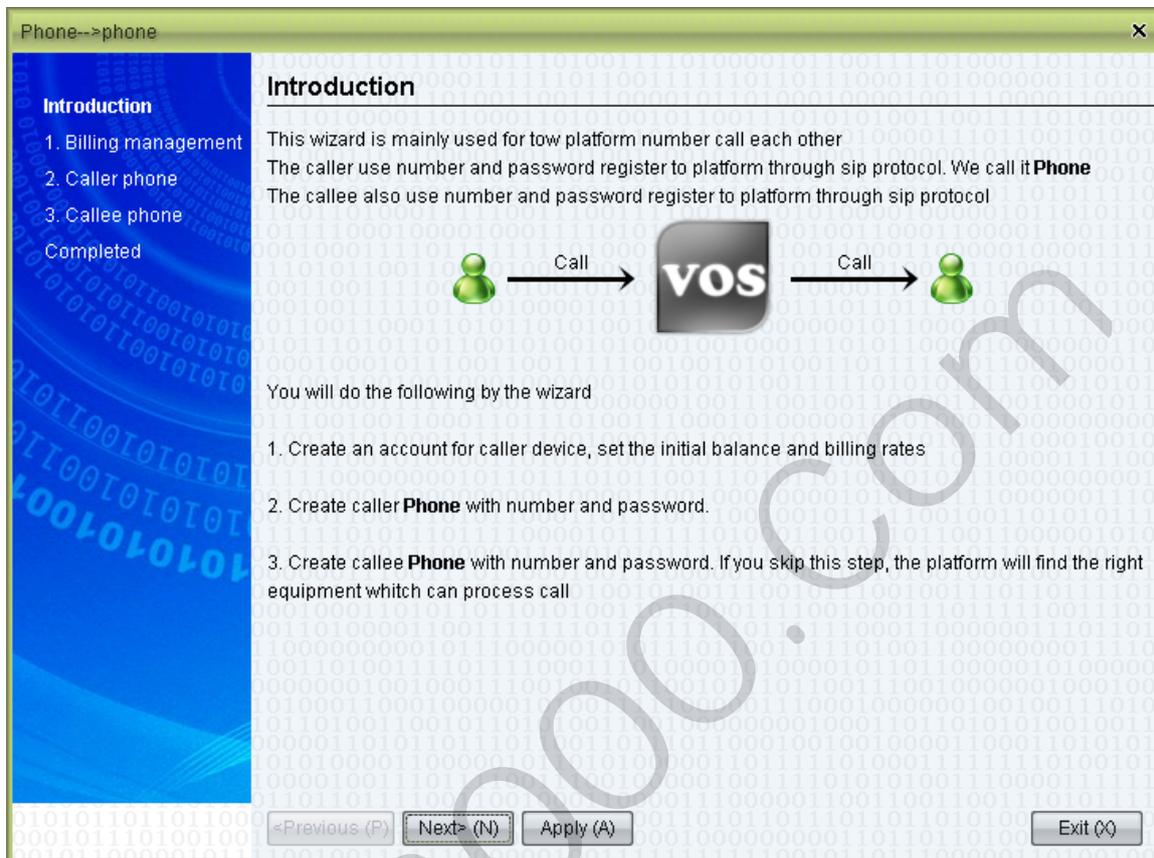


You will do the following by the wizard

1. Create an account for caller device, set the initial balance and billing rates
2. Create caller **Mapping gateway** with its static IP address
3. Create callee **Phone** with number and password. If you skip this step, the platform will find the right equipment which can process call

<Previous (P) Next> (N) Apply (A) Exit (X)

Phone → phone



Phone → routing gateway

Phone-->routing gateway

Introduction

1. Billing management
2. Caller phone
3. Routing gateway
Completed

Introduction

This wizard is mainly used for platform number call outside numbers
The caller use number and password register to platform through sip protocol. We call it **Phone**
The callee device has a static ip and numbers is often not fixed. We call it **Routing gateway**

Call

VOS

Call

You will do the following by the wizard

1. Create an account for caller device, set the initial balance and billing rates
2. Create caller **Phone** with number and password.
3. Create callee **Routing gateway** with its static IP address. If you skip this step, the platform will find the right equipment which can process call

<Previous (P) Next> (N) Apply (A) Exit (Q)

2.16 Tools

2.16.1 Customer Fee Rate Automatically Create

This function is used to create fee rate automatically.

Customer fee rate automatically create

Input fee rate

Filter

Rate prefix

Area prefix

Rate type Net

Area name

Base fee rate

Supplier fee rate

Desired profit 0.000 %

Generate fee rate

Customer fee rate

Compare files

Generate Off

How to Start

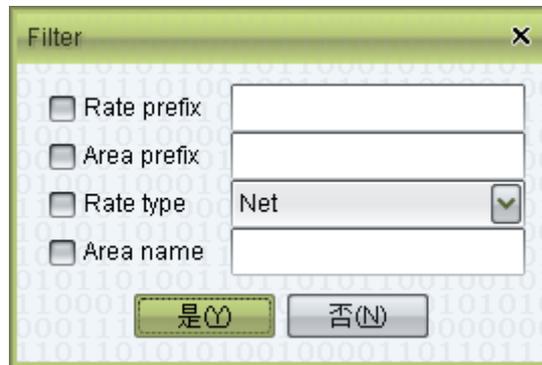
Double-click “Navigation > Tools > Customer fee rate automatically create”

Table Items

- Rate prefix
- Area Prefix
- Rate type
- Area name
- Base fee rate:
- Supplier fee rate:
- Desired profit
- Customer fee rate
- Compare files

2.16.2 Compare Fee Rate Group Minute Cost

This function is used to compare fee rate group.



How to Start

Double-click "Navigation > Tools > Compare fee rate group minute cost"

Table Items

- Rate prefix
- Area prefix
- Rate type
- Area name

2.17 Other Operation

2.17.1 Debug Trace

This function is used to track call signaling.



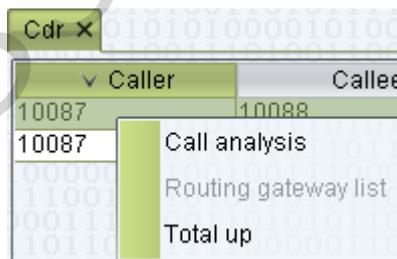
How to Start

- Click “System > Debug trace”

Usage

- Check “On” to enable track, uncheck to disable.
- Trace length: calculate from current time, stop track when timeout. Uncheck to track all the time.

After enable track, right click in “Current call” or “Cdr” to see call analysis.



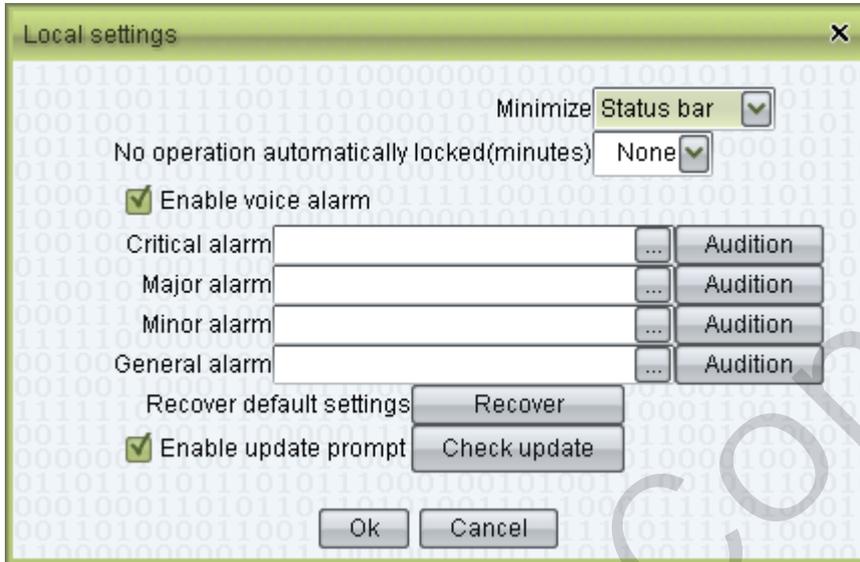
NOTE

System uses 2 files to record track signaling, so the actual space will be double. If one file over the file size limit, system will use the other file.

All track signaling will be saved, unless file has been covered.

2.17.2 Local Settings

This function is used to configure client settings.



How to Start

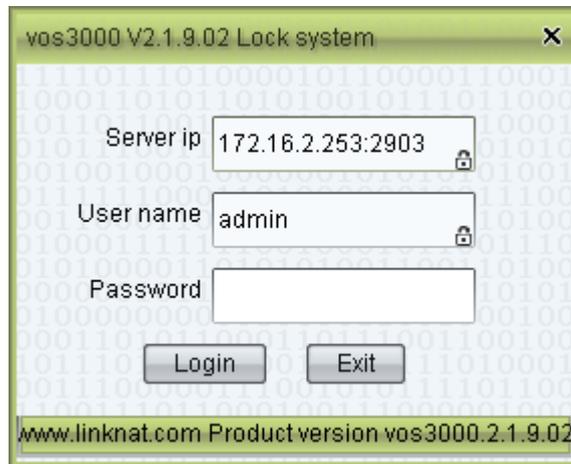
- Click “System > Local settings”

Usage

- Minimize: status bar/System tray
- No operation automatically locked
- Enable voice alarm: customer alarm audios.
- Recover default settings
- Enable update prompt: click to enable.

2.17.3 Lock System

This function is used to hide client interface.

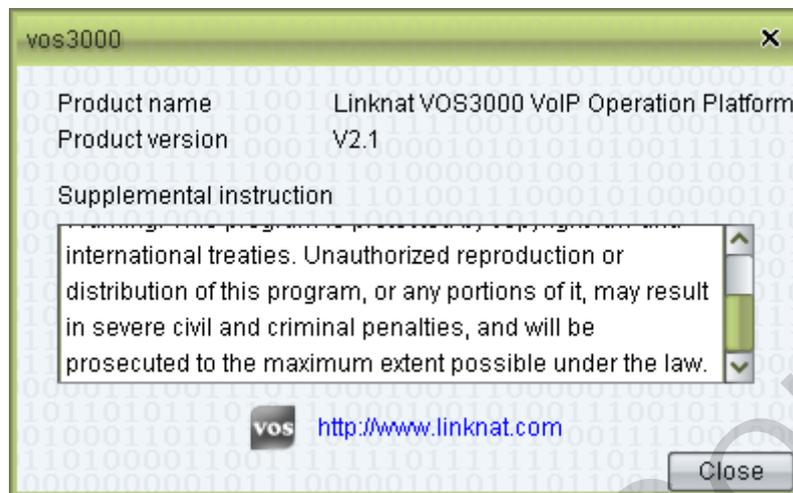


How to Start

- Click "System > Lock system"

2.17.4 Product Instruction

This function is used to show product instruction.



How to Start

- Click "System > Product instruction"

3 Functional scenarios

About This Chapter

This chapter introduces the typical functional scenarios of vos3000.

3.1 First Usage

3.1.1 Whole Sales

Operation

- Create Fee Rate Group
- Create Fee Rate
- Create Account
- Create Mapping Gateway
- Create Routing Gateway

Example

- Customer IP: 172.16.1.11
- Routing Gateway IP: 172.16.1.12

Double click “Navigation > Rate management”, add rate group 0.5 and apply.



Double click “Number of rates > 0” to add fee rate.

Rate management		Rate<0.05>management x		
Rate prefix	^ Area prefix	Rate type	Area name	Billing rate
	25	Domestic		0.0500000

Double click “Navigation > Account management”, add account test and apply.

Account management ×				
^ Account id	Account name	Current balance	Overdraft limit	Billing rate
test	test	12,221.970	0.000	0.05

Double click “Number of gateway > 0” to add mapping gateway.

Account management		Account<test>gateway management ×		
^ Gateway id	Lock type	Authorization type	Gateway group	Line limit
caller.128	No lock	Domestic		30

Double click “Navigation > Routing gateway”, add routing gateway caller.125 and apply.

Routing gateway ×					
^ Gateway id	Gateway prefix	Prefix mode	Gateway group	Lock type	Line limit
callee.125	9	Continual		No lock	30

After configuration, ask customer to send SIP call, then open “Navigation > Data query > Cdr” to see call result.

3.2 The Pickup Call Transfer

3.2.1 Usage Scenarios

- When A calls B, B needs to change another device to answer, so use this function to set the transfer access code, hang up, and arrive at another phone to continue the call.

3.2.2 Position

- Softswitch system parameter SS_CALL_TRANSFER_WAIT_ACCESS_KEY
- Softswitch system parameter SS_CALL_TRANSFER_WAIT_ACCESS_TIMEOUT

3.2.3 Association Settings

- It is necessary to add the key to the comma interval of the system parameter - toll free special service number (generating bill), otherwise, the transfer will fail because the callee cannot be charged.
- Transfer confirmation key SS_CALL_TRANSFER_END_KEY

3.2.4 Functional Principle

- A call B, B1 after answering, press “generation call transfer start key” + transfer access code + “transfer confirmation key”, after pressing, B1 will automatically hang up the phone. B use another telephone B2 to press “substitute call transfer start key” + dial key, and then the phone can be picked up after entering the transfer access code according to the prompt.

4 Supplement

About This Chapter

This chapter introduces the characteristics of VOS3000 interface.

4.1 Function Explanation

4.1.1 Network Routing Quality Reserve Time

Position:

- Softswitch management-Additional settings-System parameter-SS_GATEWAY_QUALITY_RESERVE_SEPARATE.
- Softswitch management-Additional settings-System parameter-SS_GATEWAY_QUALITY_RESERVE_TIME.

Functional Principle:

- According to this setting, the system divides the ASR calculation into several time periods, the length of each time period is $\frac{SS_GATEWAY_QUALITY_RESERVE_TIME}{SS_GATEWAY_QUALITY_RESERVE_SEPARATE}$
- For example, if SS_GATEWAY_QUALITY_RESERVE_SEPARATE is 10, SS_GATEWAY_QUALITY_RESERVE_TIME is 600, then the ASR of each time period is 60 seconds, and the ASR of a certain time point is the mean value of the ASR of the last 10 segments.

4.1.2 NAT Keep

Position:

- Softswitch SIP parameter SS_SIP_NAT_KEEP_ALIVE_SEND_ONE_TIME, SS_SIP_NAT_KEEP_ALIVE_SEND_INTERVAL, SS_SIP_NAT_KEEP_ALIVE_PERIOD, SS_SIP_NAT_KEEP_ALIVE_MESSAGE

Softswitch<MBX3000>settings		
SIP Parameter H323 Parameter System parameter		
^ Parameter name	Parameter value	Parameter description
SS_SIP_AUTHENTICATION_CODE	Unauthorized(401)	Error code when sip auth failure
SS_SIP_AUTHENTICATION_RETRY	6	Retry times for sip auth failure 0-10
SS_SIP_AUTHENTICATION_TIME...	10	Time for sip authentication 3-60
SS_SIP_E164_DISPLAY_FROM	Ignore	Mode of sip display information
SS_SIP_NAT_KEEP_ALIVE_MESS...	HELLO	Content of nat keep message
SS_SIP_NAT_KEEP_ALIVE_PERI...	30	Nat keep message's period(seconds) 10-86400
SS_SIP_NAT_KEEP_ALIVE_SEND...	500	Nat keep message's send interval(million seconds) ...
SS_SIP_NAT_KEEP_ALIVE_SEND...	5000	Nat keep message's quantity per time 100-1000000

Usage scenarios:

- In normal device registration, the registration is maintained by the device REGISTER. When the device does not support REGISTER keeping, the vos3000 can send UDP messages to keep the NAT channel.

Functional Principle:

- Nat Keep Message Content:
If it is not set, the server will not send heartbeat message. Set the content, such as Hello, then the heartbeat message sent by the server is hello.
- Nat Keep Message Sending Period (SEC):
When the UDP heartbeat messages of all NAT devices cannot be sent within this cycle, the system will send heartbeat messages again from the beginning when the cycle arrives, which may cause some devices to fail to receive heartbeat messages.

4.1.3 SIP Timer Protocol

Position:

- Softswitch SIP parameter SS_SIP_SESSION_TIMEOUT_EARLY_HANGUP, SS_SIP_SESSION_TTL, SS_SIP_SESSION_UPDATE_SEGMENT

Softswitch<MBX3000>settings		
SIP Parameter H323 Parameter System parameter		
^ Parameter name	Parameter value	Parameter description
SS_SIP_SESSION_TIMEOUT_EAR...	0	Sip timer no update early hangup(seconds) 0-60
SS_SIP_SESSION_TTL	600	Sip timer protocol expire(seconds) 90-14400
SS_SIP_SESSION_UPDATE_SEG...	2	Sip timer update interval 2-10

Usage scenarios:

- In the case of abnormal network, it is used to detect the existence of calls and avoid the generation of ultra long bills.

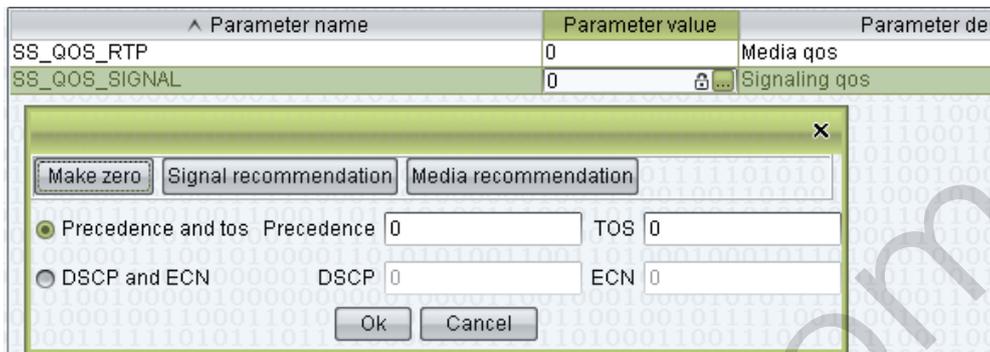
Functional Principle:

- Please refer to RFC protocol for details.

4.1.4 Signaling QoS

Position:

- Operation management-Softswitch management-System parameter SS_QOS_SIGNAL, SS_QOS_RTP



Usage scenarios:

- In order to meet the requirements of the switch equipment to transmit the message in accordance with the specific type, and ensure the voice call quality.

Functional Principle:

- The QoS field of IP header in the message is Differentiated Services Field: 0xa0 (DSCP: Unknown, ECN: ECT(1)), a0 in 0xa0 is the hexadecimal of the value set by the vos3000 parameter.

Application results:

- Message Signaling Part:

No.	Time	Source	Destination	Protocol	Length	Info
55	2018-05-23 15:06:56.061613	172.16.5.30	172.16.5.35	SIP/SDP	604	Request: INVITE sip:172.16.5.35
56	2018-05-23 15:06:56.061900	172.16.5.35	172.16.5.30	SIP	311	Status: 100 Trying
57	2018-05-23 15:06:56.066891	172.16.5.35	172.16.5.30	SIP/SDP	754	Request: INVITE sip:8002@172.16.5.30:51537
58	2018-05-23 15:06:56.069414	172.16.5.30	172.16.5.35	SIP	353	Status: 100 Trying

▶ Frame 57: 754 bytes on wire (6032 bits), 754 bytes captured (6032 bits)						
▶ Ethernet II, Src: Vmware_3a:80:a8 (00:0c:29:3a:80:a8), Dst: HewlettP_39:bb:76 (e8:39:35:39:bb:76)						
▶ Internet Protocol Version 4, Src: 172.16.5.35, Dst: 172.16.5.30						
0100 = Version: 4						
.... 0101 = Header Length: 20 bytes (5)						
▶ Differentiated Services Field: 0xa0 (DSCP: CS3, ECN: Not-ECT)						
Total Length: 740						
Identification: 0x0000 (0)						

0000	e8 39 35 39 bb 76 00 0c 29 3a 80 a8 08 00 45	80	95 9 v . .) : . . . E
0010	02 e4 00 00 40 00 fd 11 18 47 ac 10 05 23 ac 10		. . . @ . . . G . . . # . . .
0020	05 1a 13 c4 c9 51 07 d0 65 d3 49 da 56 49 54 45	 0 . . . a c T M V T F

- Message media part:

```

55 2018-05-23 15:06:56.061613 172.16.5.30 172.16.5.35 SIP/SDP 604 Request: INVITE sip:172.16.5.35 |
56 2018-05-23 15:06:56.061900 172.16.5.35 172.16.5.30 SIP 311 Status: 100 Trying |
57 2018-05-23 15:06:56.066891 172.16.5.35 172.16.5.30 SIP/SDP 754 Request: INVITE sip:8002@172.16.5.3..
58 2018-05-23 15:06:56.069414 172.16.5.30 172.16.5.35 SIP 353 Status: 100 Trying |
59 2018-05-23 15:06:56.071795 172.16.5.30 172.16.5.35 SIP 354 Status: 180 Ringing |
60 2018-05-23 15:06:56.071912 172.16.5.35 172.16.5.30 SIP 413 Status: 180 Ringing |
114 2018-05-23 15:06:57.343038 172.16.5.30 172.16.5.35 RTP 214 PT=ITU-T G.711 PCMU, SSRC=0x98CB022..
115 2018-05-23 15:06:57.343071 172.16.5.35 172.16.5.30 RTP 214 PT=ITU-T G.711 PCMU, SSRC=0x98CB022..
116 2018-05-23 15:06:57.345786 172.16.5.30 172.16.5.35 SIP/SDP 549 Status: 200 OK |
117 2018-05-23 15:06:57.345966 172.16.5.35 172.16.5.30 SIP 351 Request: ACK sip:8002@172.16.5.30:5..
118 2018-05-23 15:06:57.346248 172.16.5.35 172.16.5.30 SIP/SDP 629 Status: 200 OK |

> Frame 115: 214 bytes on wire (1712 bits), 214 bytes captured (1712 bits)
> Ethernet II, Src: Vmware_3a:80:a8 (00:0c:29:3a:80:a8), Dst: HewlettP_39:bb:76 (e8:39:35:39:bb:76)
+ Internet Protocol Version 4, Src: 172.16.5.35, Dst: 172.16.5.30
  0100 .... = Version: 4
  .... 0101 = Header Length: 20 bytes (5)
  > Differentiated Services Field: 0xa0 (DSCP: CS5, ECN: Not-ECT)
  Total Length: 200

0000 e8 39 35 39 bb 76 00 0c 29 3a 80 a8 08 00 45 a0 .959-v...):...E
0010 00 c8 3e fd 00 00 fd 11 1b 26 ac 10 05 23 ac 10 >.....&...#...
0020 05 1e 27 14 27 10 00 b4 63 27 80 80 00 01 00 00 .'. . . . c' . . . .
0030 00 a0 98 cb 02 26 ff . . . . & . . . .

```

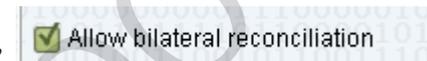
4.1.5 Enable bilateral reconciliation

Position:

“Routing gateway-Additional setting-Others”



“Mapping gateway-Additional setting-Others”



Parameter :

Parameter name	Parameter value	Parameter description
SERVER_MAIL_CUSTOMER_BILL	Off	Smtip send account consumption email automatically
SERVER_MAIL_SMTP_FROM	127.0.0.1	Smtip domain name
SERVER_TRACE_MASK	ERROR	Set display of debug information
SERVER_GATEWAY_ROUTE_BILATERAL_RECONCILIATION_LI	500	Routing gateway bilateral reconciliation simultaneously limit 1-10000
SERVER_GATEWAY_BILATERAL_RECONCILIATION_PERIOD	30	Routing gateway bilateral reconciliation period (minutes) 10-10080

SERVER_GATEWAY_ROUTE_BILATERAL_RECONCILIATION_LINE: Routing gateway bilateral reconciliation simultaneously limit 1-10000

SERVER_GATEWAY_BILATERAL_RECONCILIATION_PERIOD: Routing gateway bilateral reconciliation period (minutes) 10-10080

Functional Principle:

This function should be enabled as much as possible when both parties are VOS3000/VOS5000. This function realizes real-time account reconciliation between two trusted platforms. Through the alarm mechanism, it prevents large accounting deviations between platforms. It can effectively prevent any charging inconsistencies caused by incorrect rate configuration and network signaling errors. Avoid accounting disputes to the greatest extent. Because the system alarm mechanism is designed to have alarm capabilities on both platforms for account reconciliation, that is, as long as one party confirms that there is a deviation in the accounting, an alarm will be generated on that party's platform, thereby ensuring that even if one of the platforms is faulty, it can effectively prevent accounting disputes.

Usage scenarios:

Step 1 Configuration of platform to initiate reconciliation

“routing gateway” > “additional setting” > “other” > “enable bilateral

reconciliation” > Enable bilateral reconciliation

Step 2 Enable alarm setting on platform which initiate reconciliation

Begin time	End time	Gateway name	Alarm type	Alarm severity	Upper	Lower	Period	Voice alarm
00:00:00	24:00:00	test11	Routing bilateral reconciliation ...	Minor	1	-1	Default	Default

Step 3 Configuration of platform to receive reconciliation

“mapping gateway” > “additional setting” > “other” > “allow bilateral

reconciliation” > Allow bilateral reconciliation

Step 4 Enable alarm setting on platform which receive reconciliation

Begin time	End time	Gateway name	Alarm type	Alarm severity	Upper	Lower	Period	Voice alarm
00:00:00	24:00:00	对接vos5000	Mapping bilateral reconciliation deviation	Minor	2.000	-1.000	Default	Default

Step 5 Perform “Bilateral reconciliation” of routing gateway on platform to initiate reconciliation

Gateway name	Route pr...	Number prefix	Prefix mode	Gateway group	Lock type	Line limit	Priority	Softswit
test11	4.000		Continual		No lock	30	1	All

- Current call
- Routing gateway analysis >
- Alarm monitor
- Total gateway
- Bilateral reconciliation**

Gateway name	Routing clearing account id	Softswitch	Local balance	Remote balance	Routing bilateral reconciliation deviation	Status
test11	test11	MBX5000_172.16.4.182	97.000	98.600	1.600	

- Bilateral reconciliation
- Balance calibration

- **Bilateral reconciliation:** Get the reconciliation results of bilateral accounts
- **Balance calibration:** Calibrate the balance, the local balance will be automatically synchronized with the remote balance

Gateway name	Routing clearing account id	Softswitch	Local balance	Remote balance	Routing bilateral reconciliation deviation	Status
test11	test11	MBX5000_172.16.4.182	98.600	98.600	0.000	

4.2 Interface Description

4.2.1 Filters

The wildcard characters “*” and “?” can be used to specify filter criteria. For example, the filter criterion “800*” indicates all strings starting with “800”, and the filter criterion “888??00” represents all 7-digit strings that start with “888” and end with “00”.



NOTE

Use “*” and “?” to filter themselves, other characters have no transferred meaning.

The time of filter criteria in CDR and related spreadsheets can be specified according to either “Beginning of the call” or “End of the call”. When “Beginning of the call” is specified, then calls started in the specified time span will be matched. Otherwise, calls ended in the specified time span will be matched. Usually, operators use the time of the “End of the call” to classify calls.

4.2.2 Shortcuts

F5: Enable filtering

CTRL + C: Copy the selected table cells

ALT + F: Open “Rate management”

ALT + K: Open “Shortcuts”

ALT + S: Open “Package management”

ALT + D: Open “Mapping gateway”

ALT + G: Open “Routing gateway”

ALT + C: Open “Account management”

ALT + P: Open “Phone management”

ALT + A: Open “Current call”

ALT + H: Open “Cdr”

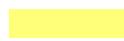
ALT + U: Open “User management”

ALT + L: Open “System log”

4.2.3 Cell Colors

Colors of Table Cell

White: Normal

: To be added after clicking the “apply” button

: To be modified after clicking the “apply” button

: To be deleted after clicking the “apply” button

-  : Selected
-  : Operation failed

Colors of Gateway Table Cell

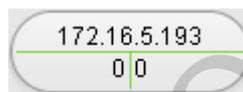
-  : Bar outgoing/incoming calls
-  : Bar all calls
-  : Conflict IP address configurations for mapping gateways

Colors of Account Table Cell

-  : Insufficient balance
-  : Account terminated

4.2.4 Floating Icon

users with the query authority except agent, will see the floating icon as following:



Top: server ip

Left bottom figure: shows current concurrency

Right bottom figure: shows queue length of cdr to be processed

4.3 System Description

4.3.1 Dial Plan

Original Prefix: prefix before change dial plan.

Target Prefix: prefix after change dial plan.

Table 4-1 Dial Plan Description

Original Prefix	Target Prefix	Number	Number after Change Dial Plan	Memo
0	0	02584316146	02584316146	do not change dial plan prefix 0
010	025	01012345678	02512345678	
025		02584316146	84316146	remove prefix 025
*	025*	117	025117	add prefix 025
025*	010	02584316146	010	change number with prefix 025 into 010
	010*	02584316146	01002584316146	add prefix 010
*		02584316146		delete all number
*	12345678	02584316146	12345678	change all number into 12345678
025*	8008100-8008121	02584316146	8008100 or 8008101 or...	change number into one of 8008100 to 8008121
*	12345?78	02584316146	12345178 or 12345278 or...	? will become a random number
*	12345678;8008100-8008121;12345?78	02584316146	12345678 or number between 8008100 and 8008121 or 12345?78	randomly choose one of the rules, then change dial plan

Original Prefix	Target Prefix	Number	Number after Change Dial Plan	Memo
0??8431	8431	02584316146	84316146	for original prefix, there can be any two digits between 0 and 8
0134	\$\$\$	0134131	013131	\$ means keep and not change dial plan the position

Multiple substitution targets can be specified, separated by the symbol “,”.

When the symbol “?” exists in the substitution targets, each “?” will be replaced by randomly generated digit. The “?” can occur more than once in the target pattern.

When multiple Dial Plans exist, the longest matching pattern will be selected. For example, if there are two Dial Plans: one replaces “0” with “0”, while the other replaces “010” with “025”, then the number “01012345678” will be replaced with “02512345678” (since “010” is longer than “0”).

The pattern “*” has the lowest priority and will only be matched when there are no other matching patterns.

Escape character: L, E, G

L: if character behind L is a number, means the number length will shorter than the specified value. For example: 13L9 represents all 11-digit numbers that start with 13 prefix

E: if character behind L is a number, the number length will equal the specified value.

For example: 010G7 represents all 10- digit numbers that start with 010 prefix

F: if character behind F is a number, the number length will longer than the specified value.

For example: 010G7 represents all 10- digit numbers that start with 010 prefix

Note : escape character with backslash before(/L,/E,/G) has no transferred meaning.

4.3.2 Media Proxy

Parameters below can be set in “Operation management > Softswitch management > Additional settings > System parameter”.

See SS_MEDIAPROXYMODE:

- On: Enable Media Proxy.
- Off: Disable Media Proxy.
- Auto: decide by softswitch, see below.
- Must On: Forced Enable Media Proxy.

Step 1 If caller or callee must on, then enable

Step 2 If caller or callee disabled media proxy, then disable.

- Step 3** If caller or callee enabled media proxy, then enable.
- Step 4** If callee enabled local ring, then enable.
- Step 5** If phone and gateway use dynamic register and encrypt, then enable.
- Step 6** If SS_MEDIAPROXYBETWEENNET is on, caller and callee from different network, then enable.
- Step 7** If SS_MEDIAPROXYBEHINDNAT is on.
If phone and gateway in the same NAT, SS_MEDIAPROXYSAMENAT is on, then enable.
If phone and gateway in different NAT, one of them in private network, then enable.
- Step 8** Finally, disable.
----End

4.3.3 Routing Gateway Sorting

- Step 1** After the "routing first / second routing strategy" is enabled for the mapping gateway or the calling phone, the next step will be taken if there is no such policy
- Step 2** According to the longest prefix matching principle, the route with the longest prefix takes precedence
- Step 3** Prefix priority of routing gateway
- Step 4** Routing gateway priority (the smaller the higher)
- Step 5** Sort by Line Usage
If SS_GATEWAYASRROUTESORTCONFIG is Before Current Day Total Call, then sort by route's ASR. Routings which disabled real-time computing ASR priority than enabled one.
If SS_GATEWAYFEERATEROUTESORTCONFIG is Before Current Day Total Call, then sort by routing gateway's Lowest Rate per Second. Gateway which disabled Sort by Lowest Rate per Second is prior than enabled.
If the above 2 parameter values are the same, then sort by SS_GATEWAYFEERATEROUTEBEFOREASR.
- Step 6** Sort by Current Day Total Call
If SS_GATEWAYASRROUTESORTCONFIG is Before Gateway ID, then sort by route's ASR. Routings which disabled real-time computing ASR priority than enabled one.
If SS_GATEWAYFEERATEROUTESORTCONFIG is Before Gateway ID, then sort by routing gateway's Lowest Rate per Second. Gateway which disabled Sort by Lowest Rate per Second is prior than enabled.
If the above 2 parameter values are the same, then sort by SS_GATEWAYFEERATEROUTEBEFOREASR.
- Step 7** Sort by Gateway ID
----End



NOTE

After open routing gateway additional settings "Calculate routing quality in real time" or "Sort by lowest rate per second", will recording to "Softswitch management - Additional settings - System parameter", insert Step5 to Step7

4.3.4 Package Billing Order

Use Free Duration.
See time span of the free duration, the longer the later.

See invalid time, the earlier the prior.
See effective date, the earlier the prior.

- Step 1** Use Free Money Amount
See rent type, prior is “Day > Month > Year”.
See invalid time, the earlier the prior.
See effective date, the earlier the prior.

- Step 2** Use Account Money
----End

4.3.5 Parameter Description

4.3.5.1 VOS3000 Parameter

Parameter Name	Default Value	Parameter Description
EXTERNAL_WEB_SEND_PHONE_ONLINE	Off	Interface: phone online/offline transfer
EXTERNAL_MAX_CDR_PENDING_SIZE	10000	Interface: queue size of resend cdr 1000-100000
EXTERNAL_SEND_CDR	Off	Interface: send cdr
SERVER_ALARM_CUSTOMER_BALANCE_MAX_SIZE	1000	Number of accounts in “Navigation > Alarm management > Alarm settings > Balance alarm”
SERVER_ALARM_DATABASE_IGNORE_ERROR_CODE		Database error code without warning
SERVER_ALARM_DISABLE	Off	Off: enable alarm. On: disable alarm.
SERVER_ALARM_E164S		Default e164 setting in “Navigation > Alarm management > Alarm Setting”
SERVER_ALARM_EMAIL		Default email setting in “Navigation > Alarm management > Alarm Setting”
SERVER_ALARM_EMAIL_DELAY	300	Unit: second. Interval for email alarm.
SERVER_ALARM_ENABLE_EMAIL	Off	Default email alarm setting in “Navigation > Alarm management > Alarm Setting”
SERVER_ALARM_ENABLE_VOICE	Off	Voice alarm

Parameter Name	Default Value	Parameter Description
SERVER_BILLING_CUSTOMER_DEFAULT_FEE_RATE		Default fee rate group for HW interface
SERVER_BILLING_FEE_PRECISION	0.0000000	Billing money accuracy 0-1000
SERVER_BILLING_FEE_UNIT	0.0000000	Billing money unit 0-1000
SERVER_BILLING_FORWARD_PREFIX		Billing prefix for Call Transfer
SERVER_BILLING_FREE_E164S		Service Number for free, no time limit
SERVER_BILLING_FREE_TIME	0	Unit: second. Charged Duration will cut this value, but won't change conversation time.
SERVER_BILLING_GATEWAY_ROUTE_PREFIX		Routing gateway additional prefix, e.g. set 99,88 then if callee is 99123, will be charged as 123 in CDR
SERVER_BILLING_HOLD_TIME_PRECISION	1000	Unit: million second. Time precision, e.g. set 50, if duration is 21.049s, then uses 21s, else if duration is 21.050, then uses 22s.
SERVER_BILLING_NO_CDR_E164S		See SERVER_BILLING_FREE_E164S, but these numbers won't create CDR
SERVER_BILLING_PREVENT_OVERDRAFT_ADVANCE_TIME	1	Account anti overdraft advance the number of minutes each time (minutes) 1-15
SERVER_BILLING_PROFIT_CALCULATE	<Call charges>Sub<Call expense>	Call profit calculation
SERVER_BILLING_RECOR_SERVER_HANG_UP	On	Valid when SERVER_BILLING_RECOR_ZERO_HOLD_TIME is on. If SERVER_BILLING_RECOR_SERVER_HANG_UP is off, hold time is 0 and hang up by server, the CDR won't be saved.

Parameter Name	Default Value	Parameter Description
SERVER_BILLING_RECO DR_ZERO_HOLD_TIME	On	If system is under attack, turn off this to reduce process load
SERVER_BILLING_RECO RD_ILLEGAL_CALL	On	Illegal call: caller's IP cannot find in mapping gateways and caller's number cannot find in phones.
SERVER_BILLING_USE_ ROUTING_GATEWAY_P REFIX		
SERVER_CALL_HELPER_ E164		Call helper number, can be used with SERVER_BILLING_FREE _E164S
SERVER_CDR_FILE_WRI TE_INTERVAL	None	Additional write call record file, the new file created time interval (seconds) 60-86400
SERVER_CDR_FILE_WRI TE_MAX	2048	Additional write call record file, the maximum number of reserved file 10-4096
SERVER_CDR_REAL_TI ME_REPORT_SERVER		Additional send call record to server address
SERVER_DATABASE_VE RSION		Version of data
SERVER_DISPLAY_CHA RT_GATEWAY_SIZE	10	Gateway number in "Gateway Performance" and "Period Connect analysis"
SERVER_DISPLAY_CUST OMER_ALARM_MONEY	20	In "account management", if current balance below the value, color of the row will be changed
SERVER_DISPLAY_MON EY_PRECISION	3	Money Precision, e.g. set value 3, then 1 will be shown as 1.000.
SERVER_DNS_UPDATE_I NTERVAL	600	Unit: second. Used for "Domain management"
SERVER_GATEWAY_RO UTE_PREFIX		Routing additional routing prefix, separated by commas.
SERVER_IPPBX_DEFAUL T_LANGUAGE	chinese	Default language of ip pbx service

Parameter Name	Default Value	Parameter Description
SERVER_LOGIN_FAILED_DISABLE_TIME	120	Time of disable user login when failed several times(seconds) 30-7200
SERVER_E164_NON_STANDARD_PREFIX		If caller is phone and SERVER_E164_INTERNATIONAL_CALLEE_REBUILD is on, before routing, callee number will omit this prefix. This parameter won't change callee number.
SERVER_MAIL_CUSTOMERE_BILL	Off	SMTP Send Account Consumption Email Automatically. "Account management > Customer Information > Email"
SERVER_MAIL_REPORT_DAY_OF_WEEK	Monday	"Account management > Customer Information > Email > Report Sending Mode > Weekly"
SERVER_MAIL_SMTP_FROM	127.0.0.1	Email sender's IP
SERVER_MAIL_SMTP_FROM_USER	VOS3000 Limiteddemo	Smtpt sender
SERVER_MAIL_SMTP_SERVER	127.0.0.1	SMTP Server IP
SERVER_MAIL_SYSTEM_NOTICE		System notification email address
SERVER_MAX_CDR_PENDING_LIST_LENGTH	100000	Length of cdr queue limit 10000-100000
SERVER_MAX_INFO_ON_E_MESSAGE	10000	Max Send Number per time when Data Changed
SERVER_MGC_EQUIPMENT_IPS		Reliable equipment ip
SERVER_MGC_EXTERNAL_IPS		External interface device ip
SERVER_NTP_SERVER	time-a.nist.gov	Network time server (sntp)
SERVER_PASSWORD_LENGTH	8	Default Length of Password
SERVER_PASSWORD_TERMINAL_ADDITIONAL_CHARACTERS		Additional characters for phone and gateway random passwords. Default: 0-9

Parameter Name	Default Value	Parameter Description
SERVER_PAY_DELAY_CUSTOMER_EXPIRE_DAY	365	Unit: day. Extend the validity after recharge
SERVER_PAY_NORMAL_EXPIRE_ACTIVE_DISCOUNT	0	Deduct Original Amount Percent, when Normal Recharge to Expired Account(%)
SERVER_PAY_NORMAL_EXPIRE_ACTIVE_FEE	0.0	Deduct Fee per Day, when Normal Recharge to Expired Account(Dollar)
SERVER_PAY_PHONE_CARD_EXPIRE_ACTIVE_DISCOUNT	0	Deduct Original Amount Percent, when use phone card to Recharge Expired Account(%)
SERVER_PAY_PHONE_CARD_EXPIRE_ACTIVE_FEE	0.0	Deduct Fee per Day, when use phone card to Recharge Expired Account(Dollar)
SERVER_PAY_PHONE_CARD_PAY_RELATE_TO_AGENT	Off	Charge Agent Accounts, which Use Lower Rate, at the same time, when customers use Phone Card to do charge.
SERVER_PHONE_AS_CALLER_MONEY_VERIFY	on	Verify callee phone account balance
SERVER_PHONE_CARD_AUTO_UNBIND_HOUR_IN_DAY	3	Checking time for phonocard account automatic unbind 0--23
SERVER_PHONE_CARD_AUTO_UNBIND_REMAIN_MONEY	None	Remain money for phonocard automatic unbind
SERVER_PHONE_CARD_CONSUMPTION_PRECISION	0.0000000	Phone card billing precision 0-1000
SERVER_PHONE_CARD_CONSUMPTION_UNIT	0.0000000	Phone card billing unit 0-1000
SERVER_PHONE_DEFAULT_LANGUAGE	1	Call Helper Default Language Code
SERVER_PSTN_PREFIX	0	PSTN Call Prefix
SERVER_QUERY_CDR_DENY_TIME		No CDR Query Time(24 hour) e.g. 18,19,20,21,22,23

Parameter Name	Default Value	Parameter Description
SERVER_QUERY_CDR_MAX_DAY_INTERVAL	31	Maximum Interval for CDR Inquiry(Day)
SERVER_QUERY_MAX_ONE_PAGE_SIZE	200000	Maximum Number of Data per Page
SERVER_QUERY_MAX_SIZE	30000000	Data Query Limit(Item)
SERVER_QUERY_NON_PAGABLE_MAX_LINES	100000	Non pagable table, maximum lines per page 1000-200000
SERVER_QUERY_ONE_PAGE_SIZE	10000	Number of Data per Page(Item)
SERVER_REPORT_AGENT_INCOME	On	Automatically generates agent income report
SERVER_REPORT_CLEARING_CUSTOMER_FEE	Off	Auto Generate Clearing Account Details Report
SERVER_REPORT_CLEARING_CUSTOMER_IO	Off	Auto Generate Account Clearing Balance Report
SERVER_REPORT_CLEARING_CUSTOMER_LOCATION_FEE	Off	Auto Generate Clearing-Account Area Details Report
SERVER_REPORT_CLEARING_GATEWAY_FEE	Off	Auto Generate Clearing Gateway Details Report
SERVER_REPORT_CUSTOMER_FEE	On	Auto Generate Revenue Details Report
SERVER_REPORT_CUSTOMER_IO	Off	Auto Generate Account Balance Report
SERVER_REPORT_CUSTOMER_LOCATION_FEE	On	Auto Generate Account Area Detail Report
SERVER_REPORT_GATEWAY_CROSS_LOCATION_ASR_ACD	Off	Automatically generate gateway cross area analysis report
SERVER_REPORT_GATEWAY_FEE	On	Auto Generate Gateway Bill Report
SERVER_REPORT_GATEWAY_MAPPING_ASR_ACD	Off	Auto Generate Mapping Gateway Connect Analysis Report
SERVER_REPORT_GATEWAY_MAPPING_LOCATION_ASR_ACD	On	Automatically generate mapping gateway area analysis report

Parameter Name	Default Value	Parameter Description
SERVER_REPORT_GATEWAY_ROUTING_ASR_ACD	Off	Auto Generate Routing Gateway Connect Analysis Report
SERVER_REPORT_GATEWAY_ROUTING_LOCATION_ASR_ACD	On	Automatically generate routing gateway area analysis report
SERVER_REPORT_PHONE_CARD_E164_FEE	On	Automatically generate bind number bill report
SERVER_REPORT_PHONE_CARD_FEE	On	Automatically generate phone card bill report
SERVER_REPORT_PHONE_FEE	On	Auto Generate Phone Bill Report
SERVER_REPORT_TIME_ZONE_LOWER		Minimum accounts time zone(milliseconds)
SERVER_REPORT_TIME_ZONE_UPPER		Maximum accounts time zone(milliseconds)
SERVER_SUPPORT_EXPIRE_NOTIFY	Off	Technical support expire reminder
SERVER_SIP_LOAD_BALANCE_SERVER		Load Balancing Server IP
SERVER_SMAP_RESERVE_TIME	None	SMAP Reserve Duration (seconds) 60-86400 None: disable.
SERVER_SOFTSWITCH_CLUSTER		IP List of Softswitch Cluster
SERVER_SOFTSWITCH_ENDPOINT_EXPIRE	3600	Unit: second. Terminal Registration Expiry Time
SERVER_SOFTSWITCH_ENDPOINT_NAT_EXPIRE	120	Unit: second. Terminal Registration Expiry Time
SERVER_SUPPORT_EXPIRE_NOTIFY	On	Technical Support Expire Reminder
SERVER_TRACE_FILE_LENGTH	40960	Size of Debug file
SERVER_TRACE_MASK	ERROR	Set Display of Debug Information
SERVER_TRACE_TO_FILE	On	Output Debug Information into File

Parameter Name	Default Value	Parameter Description
SERVER_VERIFY_CLEARING_CUSTOMER	Off	Check Callee's Phone Account On: clearing account's balance must over SERVER_VERIFY_CLEARING_CUSTOMER_REMAIN_MONEY_LIMIT.
SERVER_VERIFY_CLEARING_CUSTOMER_REMAIN_MONEY_LIMIT	0.0	Clearing Account Remain Money Limit 0-10000000
SERVER_VERIFY_CLEARING_CUSTOMER_TIME	Off	If SERVER_VERIFY_CLEARING_CUSTOMER is on, Check Clearing Account Available Time
WEB_PHONEBOOKCALLBACKACCESSNUMBER_CARD		Access Number for Web Directory billing by Phone Card
WEB_PHONEBOOKCALLBACKACCESSNUMBER_PHONE		Access Number for Web Directory billing by Phone Number

4.3.5.2 Softswitch Parameter

Table 4-2 H323 Parameter

Parameter Name	Default Value	Parameter Description
SS_H245_PORT_RANGE	10000,39999	H245 port range
SS_H323_DTMF_METHOD	H.245 alphanumeric	Default DTMF send mode
SS_H323_NUMBERING_PLAN	UnknownPlan(0)	Default value in "Routing Gateway > Additional settings > Protocol > H323"
SS_H323_NUMBER_TYPE	UnknownType(0)	Default value in "Routing Gateway > Additional settings > Protocol > H323"
SS_H323_PROGRESS_INDICATOR	ProgressInbandInformationAvailable(8)	Default value in "Mapping Gateway > Additional settings > Protocol > H323"
SS_H323_SCREENING_INDICATOR	None	Default value in "Routing Gateway > Additional settings > Protocol > H323"

Parameter Name	Default Value	Parameter Description
SS_H323_STOP_SWITCH_AFTER_OLC	Off	Default value in “Routing Gateway > Additional settings > Protocol > H323”
SS_H323_TIMEOUT_ALERTING	120	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > H323”
SS_H323_TIMEOUT_CALLPROCEEDING	20	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > H323”
SS_H323_TIMEOUT_CALLPROCEEDING_OLC	20	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > H323”
SS_H323_TIMEOUT_SETUP	5	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > H323”

Table 4-3 SIP Parameter

Parameter Name	Default Value	Parameter Description
SS_SIP_AUTHENTICATION_CODE	Unauthorized(401)	Return code for SIP authentication, when Register message or Invite message without Proxy-Authorization or Authorization.
SS_SIP_AUTHENTICATION_RETRY	6	SIP authentication retry time, when received 401 or 407
SS_SIP_AUTHENTICATION_TIMEOUT	10	Time for SIP Authentication If caller failed to get authentication within the time, Softswitch will reject the call.
SS_SIP_E164_DISPLAY_FORMAT	Ignore	Mode of sip display information
SS_SIP_NAT_KEEP_ALIVE_MESSAGE	HELLO	Content of NAT Keep Message
SS_SIP_NAT_KEEP_ALIVE_PERIOD	30	NAT Keep Message's Period(seconds) 10-86400

Parameter Name	Default Value	Parameter Description
SS_SIP_NAT_KEEP_ALIVE_SEND_INTERVAL	500	NAT Keep Message's Send Interval(million seconds)
SS_SIP_NAT_KEEP_ALIVE_SEND_ONE_TIME	3000	NAT Keep Message's Quantity per Time
SS_SIP_NO_TIMER_REINVITE_INTERVAL	7200	Maximum Conversation Time for Non-TIMER SIP Caller If SIP caller doesn't support "timer", softswitch will stop the call when the time is up.
SS_SIP_PUBLISH_EXPIRE	300	Routing gateway public update timeout default duration (seconds) 30-7200
SS_SIP_RESEND_INTERVAL	0.5,1,2,4,4,4,4,4,4,4	Resend SIP Message Interval (Second) If got no response or confirm within the time, Softswitch will resend SIP message. If exceeded the retry times, Softswitch will stop sending and regard as call failure, then try another gateway or hang up.
SS_SIP_ROUTING_GATEWAY_INVITE_USE_CONTACT	Off	Use number from request-line as callee and keep original number in To field when send invite to callee
SS_SIP_SESSION_TIMEOUT_EARLY_HANGUP	0	SIP Timer no reinvite (update) Early Hang up(Second)
SS_SIP_SESSION_TTL	600	Detecting SIP Connected Status Interval(Second) If SIP caller supports "session-timer", within the time Softswitch will detect connect status according to the retry times. If got no confirm message, Softswitch will regard as call finish, then hang up.
SS_SIP_SESSION_UPDATE_SEGMENT	2	SIP Timer reinvite (update) Interval 2--10

Parameter Name	Default Value	Parameter Description
SS_SIP_STOP_SWITCH_AFTER_SDP	On	Stop Switch Gateway After Receive SDP
SS_SIP_TIMEOUT_INVITE	10	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > SIP”
SS_SIP_TIMEOUT_RINGING	120	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > SIP”
SS_SIP_TIMEOUT_SESSION_PROGRESS	20	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > SIP”
SS_SIP_TIMEOUT_SESSION_PROGRESS_SDP	120	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > SIP”
SS_SIP_TIMEOUT_RINGING	20	Unit: second. Default value in “Routing Gateway > Additional settings > Protocol > SIP”
SS_SIP_USER_AGENT_EXPIRE	Auto Negotiation	SIP Registration Expiration Time to Other Server(seconds) 20-7200
SS_SIP_USER_AGENT_PRIVACY	Ignore	Privacy Setting for Register User Ignore: No Privacy field Id: contain Privacy: id None: contain Privacy: none
SS_SIP_USER_AGENT_RETRY_DELAY	60	Resend Interval for SIP Registration when Failed(Second) 30--600
SS_SIP_USER_AGENT_SEND_UNREGISTER	On	Send Cancel Register Message
SS_SIP_USER_AGENT_STOP_SWITCH_AFTER_INVITE_TIMEOUT	Off	Stop Switch Gateway After INVITE Timeout

Table 4-4 System Parameter

Parameter Name	Default Value	Parameter Description
SS_ACCOUNT_INDICATION_METHOD	Off	Insufficient balance alarm mode
SS_ACCOUNT_INDICATION_MONEY	10	Insufficient balance alarm threshold
SS_ACCOUNT_INDICATION_TIME	100	Insufficient remaining duration alarm threshold(minutes)
SS_AUTHENTICATION_FAILED_SUSPEND	180	Disable duration after exceed max terminal password authentication retry times(seconds) 60-3600
SS_AUTHENTICATION_MAX_RETRY	6	Max terminal password authentication retry times 0-999
SS_BLACK_LIST_CALLER_CONCURRENT_EXPIRE	86400	Malicious call dynamic caller black list expired duration(seconds)
SS_BLACK_LIST_CALLER_CONCURRENT_LIMIT	None	Malicious call dynamic caller black list concurrency limit
SS_BLACK_LIST_CALLER_MALICIOUS_CALL_LIMIT	None	Malicious call dynamic caller black list max call times
SS_BLACK_LIST_CALLER_MALICIOUS_CALL_CHECK_INTERVAL	600	Malicious call dynamic caller black list monitor cycle(seconds)
SS_BLACK_LIST_CALLER_MALICIOUS_CALL_EXPIRE	3600	Malicious call dynamic caller black list expired duration(seconds)
SS_BLACK_LIST_MALICIOUS_CALL_CHECK_INTERVAL	600	Malicious call dynamic callee black list monitor cycle(seconds)
SS_BLACK_LIST_MALICIOUS_CALL_EXPIRE	3600	Malicious call dynamic callee black list expired duration(seconds)
SS_BLACK_LIST_MALICIOUS_CALL_LIMIT	None	Malicious call dynamic callee black list max call times
SS_BLACK_LIST_NO_ANSWER_EXPIRE	2	No answer call dynamic black list expired duration(days)

Parameter Name	Default Value	Parameter Description
SS_BLACK_LIST_NO_ANSWER_LIMIT	None	No answer call dynamic black list continuous call times
SS_BLACK_LIST_NO_ANSWER_PERIODS		No answer call dynamic black list monitor period
SS_CALLCAPTURERTPORT	40000	Local Port for Call Monitor(Even Number)
SS_CALL_FAILED_INDICATION	None	Prompt phone call failed
SS_CALL_FORWARD_USING_ORIGINAL_CALLER	On	Use the Original Caller as Forward Display Number
SS_CALL_HOLD_KEY	*9	Call hold key
SS_CALL_PICK_UP_KEY	*4	Call pickup key
SS_CALL_REMAIN_TIME_INDICATION_BOUNDARY	1	Time of prompt phone remaining duration (minutes). During the call, if remain time less than the value, system will give prompt to both side.
SS_CALL_REMAIN_TIME_INDICATION	Off	Prompt phone remaining duration
SS_CALL_REPORT_IP		Call state notify address
SS_CALL_REPORT_PORT	8000	Call state notify port
SS_CALL_REPORT_RETRY	3	Call state notify retry times
SS_CALL_REPORT_RETRY_INTERVAL	6	Call state notify retry interval
SS_CALL_SERVICE_CANCEL_KEY	***	Call transfer cancel key
SS_CALL_STATE_REPORT	Off	Http call status notification
SS_CALL_TRANSFER_ASK_DISPLAY	Original caller	Ask call transfer display
SS_CALL_TRANSFER_ASK_KEY	*2	The inquisitorial call transfer start key
SS_CALL_TRANSFER_CANCEL_KEY	**	Call transfer cancel key
SS_CALL_TRANSFER_END_KEY	#	Confirm button for call transfer

Parameter Name	Default Value	Parameter Description
SS_CALL_TRANSFER_NORMAL_DISPLAY		
SS_CALL_TRANSFER_NORMAL_KEY	*1	Start button for call transfer
SS_CALL_TRANSFER_REMOTE_RING_PASS_THROUGH	On	Send color ring back tone when call transfer
SS_CALL_TRANSFER_WAIT_ACCESS_KEY	*3	The pickup call transfer start key
SS_CALL_TRANSFER_WAIT_ACCESS_TIMEOUT	120	Pickup call transfer timeout (seconds)
SS_CDR_RECORD_ILLEGAL	On	Record illegal call
SS_CDR_RECORD_NONCONNECT	Off	When saving CDR as TXT, contains CDR, which hold time is 0s
SS_CDR_RECORD_TO_FILE	Off	Save CDR as TXT
SS_DEFAULT_LOCAL_IP		Default Local Address
SS_DTMF_TIMEOUT	5	Waiting time for Automatic Call Transfer
SS_ENDPOINT_EXPIRE	300	Terminal register expire time
SS_ENDPOINT_REGISTER_REPLACE	On	Allow replace the current registered users when terminal registration.
SS_ENDPOINT_EXPIRE	3600	Terminal registration expiry time(seconds) 60-86400
SS_ENDPOINT_NAT_EXPIRE	300	Terminal registration expiry time(seconds) 60-86400
SS_ENDPOINT_REGISTER_REPLACE	On	Allow replace the current registered users when terminal registration
SS_ENDPOINTREGISTER_RETRY	6	Max retry times when terminal registration
SS_EXTERNAL_REWRITE_TYPE	None	External dial plan method
SS_ENDPOINTREGISTER_SUSPEND	180	Disable duration after exceed retry times for when terminal registration

Parameter Name	Default Value	Parameter Description
SS_ENDPOINTTIMETOLIVE	60	Interval for Lightweight Registration of Terminal(seconds)
SS_GATEWAY_ACD_CALCULATE	Off	External dial plan method
SS_GATEWAY_ACD_RESERVE_SEPARATE	10	Section for gateway's acd routing(calculated as the step size) 5-24
SS_GATEWAY_ACD_RESERVE_TIME	600	Length for gateway's acd routing(seconds) 300-86400
SS_GATEWAY_ASR_CALCULATE	Off	Real time computing asr
SS_GATEWAY_ASR_RESERVE_SEPARATE	10	Section for gateway's asr routing(calculated as the step size) 5-24
SS_GATEWAY_ASR_RESERVE_TIME	600	Length for gateway's asr routing(seconds) 300-86400
SS_GATEWAY_ASR_ROUTE_SORT_CONFIG	Before line usage	Position for routing gateway's asr routing
SS_GATEWAY_FEE_RATE_ROUTE_BEFORE_ASR	Off	Rate routing priority over asr routing
SS_GATEWAY_FEE_RATE_ROUTE_SORT_CONFIG	Before line usage	Position for routing gateway's rate routing
SS_GATEWAY_SWITCH_LIMIT	None	Times limit for Routing Gateway Auto-Switch
SS_GATEWAY_SWITCH_STOP_AFTER_RTP_START	On	Stop Switch Gateway when RTP Start
SS_GATEWAY_SWITCH_STOP_AFTER_USER_BUSY	On	Callee busy stop switch
SS_GATEWAY_SWITCH_UNTIL_CONNECT	Off	Switch Gateway Until Connect
SS_GC_INTERVAL	3600	Interval for Clean Useless Data of Softswitch(seconds)
SS_LOCAL_RING_DEFAULT	localring	Default local ring
SS_LRN_SERVER_IP		Lrn server ip
SS_LRN_SERVER_PORT	5060	Lrn server port 1-65535

Parameter Name	Default Value	Parameter Description
SS_MAPPING_GATEWAY_TIMEOUT	None	Mapping Gateway Default Timeout
SS_MAX_CALL_DURATION	None	Max Conversation Limitation(seconds)
SS_MEDIA_CHECK_TIMEOUT	2	No Media Data Hang Up Duration(minutes)
SS_MEDIA_PROXY_BEHIND_NAT	On	Forward RTP for Registered Terminals behind NAT
SS_MEDIA_PROXY_BETWEEN_NET	Off	Forward RTP for Calls between Different Networks
SS_MEDIA_PROXY_MODE	Auto	Media Proxy
SS_MEDIA_PROXY_PORT_RANGE	10000,39999	RTP Port for Forwarding Voice, use Comma to Separate the Start and End port
SS_MEDIA_PROXY_SAME_NAT	On	Forward RTP for Registered Terminals in the same NAT
SS_MOBILE_E164_LENGTH	11	Length limit for Mobile Number separated by commas (Maximum 31), uses 0 for allow all the length
SS_NON_STANDARD_PREFIX		Non standard e164 prefix, separated by commas.
SS_NO_BILLING_TO_PHONE	Off	Free billing to phone
SS_PHONE_ONLINE_REPORT	Off	Interface: phone online/offline notice
SS_PHONE_SERVICE_IP		Phone service ip
SS_PHONE_SERVICE_PORT	5062	Phone service port 1-65535
SS_REDIRECT_OFFLINE_PHONE_TO_GATEWAY	Off	If phone is offline, try to use routing gateway for routing.
SS_REDIRECT_SERVER		External billing and routing control redirect server address (Sip 3xx)
SS_REDIRECT_SERVER_AVAILABLE_TO_PHONE	Off	External billing and routing control also take effect when callee is phone

Parameter Name	Default Value	Parameter Description
SS_REPLY_UNAUTHORIZED	On	Respond to Unauthorized Registration or Call
SS_RTP_ENCRYPT_V2	XOR,RC4,AES128	Version 2 rtp encryption algorithm (XOR,RC4,AES128)
SS_SELF_SERVICE_URL	http://\$ipaddress/\$language/directlogin.jsp?name=\$loginname&password=\$password	Phone self service url
SS_TCP_CLOSE_RESET	Off	Close TCP connection in Direct Reset mode.
SS_TIMEOUT_CALL_FORWARD_NO_ANSWER	20	Time for Call Forwarding, When No Reply(seconds)
SS_TIMEOUT_PHONE_NO_ANSWER	120	Time for Hang Up, When No Reply(seconds)
SS_TRACE_CALL_FILE_SIZE	16	Call signaling trace file size limit (MB) 16-2048
SS_TRACE_FILE_LENGTH	40960	Size of Softswitch's Debug File(KB)
SS_TRACE_MASK	ERROR	Set Display of Debug Information
SS_TRACE_REGISTER_FILE_SIZE	16	Registration signaling trace file size limit (MB) 16-2048
SS_TRACE_REGISTER_MAX_TRANSMIT	5	Registration Track Maximum Data Size(MB)
SS_TRY_PROTECT_ROUTE_DELAY	None	Protect rout enable time (seconds) 0-180
SS_TRACETOFILE	On	Output Debug Information into File
SS_UNBOUND_INDICATION	Off	Prompt whether the Phone Card is Binded
SS_USE_CALLER_PHONE_DISPLAY	Off	SS_USE_CALLER_PHONE_DISPLAY
SS_VALUE_ADDED_CODECS	g729a,g729,g723,g711a,g711u	Audio codecs prior to value added (g729a,g729,g723,g711a,g711u)
SS_VALUE_ADDED_IP		Value Added Server IP
SS_VALUE_ADDED_PORT	5055	Value Added Server Port

Parameter Name	Default Value	Parameter Description
SS_VIRTUAL_IPS		DMZ Settings

4.3.5.3 Audio Service Parameter

Parameter Name	Default Value	Parameter Description
IVR_CALLBACK_KEEP_LINE_RING_TIME	5	Alerting time for callback caller reservation 0-120
IVR_CALLBACK_KEEP_LINE_TIME	30	Used for callback line keep. See IVR_CALLBACK_KEEP_LINE_RING_TIME
IVR_CALL_REPORT_IP		Send IVR second line's call state. UDP request format: Call ID, Serial Number, Call State, Caller Number, Callee Number, Forward Number, Menu ID, Menu Name. Response format: Call ID, Serial Number. Call State: Ringing(180/183)/OK/Bye
IVR_CALL_REPORT_PORT	8000	Report UDP Port
IVR_CALL_REPORT_RETRY	6	Call State Notify Retry Times
IVR_CALL_REPORT_RETRY_INTERVAL	3	Call State Notify Retry Interval
IVR_CODEC_PRIORITY	g729a,g729,g723,g711a,g711u	Voice Codecs Priority (g729a,g729,g723,g711a,g711u)
IVR_DEFAULT_ERROR_AUDIO	defaulterror	Default Error Message Voice
IVR_DEFAULT_LANGUAGE	chinese	Use Default Language when IVR don't know Client's Language
IVR_ENABLE_CARD_LINE_RESTRICT	On	Each Phone Card cannot use IVR service at the same time.
IVR_ENABLE_PARSE_INBAND	Off	Inband DTMF Analysis
IVR_ENABLE_PARSE_SECOND_LINE_INBAND	Off	Second Line Inband DTMF Analysis

IVR_MEDIA_CHECK_TIME_OUT	2	No media data hang up duration(minutes) 1-120
IVR_PARSE_DTMF_MODE	Auto	DTMF Analysis Mode
IVR_RESERVE_CALL_KEEP_TIME	300	Reservation callee number keep time(seconds) 10-1800
IVR_RINGING_TIMEOUT	120	Time for IVR Hang Up, When No Reply(seconds)
IVR_RTP_PORT	40000,47999	Media Port Range
IVR_SETUP_TIMEOUT	20	Invite Timeout Duration
IVR_SHORTE164_AUTO_RECORD_ENABLE	Off	Auto Save Number Function
IVR_SHORTE164_AUTORECORD_LENGTH	4	Auto Save Number Suffix
IVR_SHORTE164_AUTORECORD_MAX_NUMBER	5	Auto Save Number Amount
IVR_SIP_NO_TIMER_MAXIMUM_SESSION_TIME	7200	Maximum Conversation Time for Non-Timer SIP Caller
IVR_SIP_RESEND_INTERVAL	3	Interval for Resend SIP Message(seconds)
IVR_SIP_SEND_RETRY	6	Times of Resend SIP Message
IVR_SIP_SESSION_RETRY	6	Retry Times for Detecting SIP Connected Status
IVR_SIP_SESSION_TTL	600	Interval for Detecting SIP Connected Status(seconds)
IVR_SOFTSWITCH_IP		IP of Softswitch
IVR_SOFTSWITCH_SIP_PORT	5060	Port of SIP Softswitch
IVR_SOFTSWITCH_AVAILABLE_IP		Access IP List, separated by commas
IVR_TRACE_FILE_SIZE	40960	Size of Softswitch's Debug File(KB) 4096-4096000
IVR_TRACE_MASK	ERROR	Set Display of Debug Information
IVR_TRACE_TO_FILE	On	Output Debug Information into File
IVR_WEB_CALLBACK_SAMPLE_TIME_CODEC	g729a	Codec for Call Both Side

IVR_ALARM_CALLER_E164		Voice Alarm Caller Number
IVR_ALARM_CONFIRM_KEY		Voice Alarm Confirm Key
IVR_ALARM_PERIOD	5	Voice Alarm Period(minutes)
IVR_ALARM_PRE_AUDIO	alarmpreaudio	Voice Alarm Pre-Prompt Audio
IVR_ALARM_RETRY	6	Voice Alarm Retry Times
IVR_ALARM_RETRY_INTERVAL	20	Voice Alarm Retry Interval
IVR_VOICEMAIL_EXPIRE_DAY	7	Voice Mail Preservation Days
IVR_VOICEMAIL_MAX_NUMBER	10	Voice Mail Max Items
IVR_VOICEMAIL_MAX_TIME	60	Voice Mail Recording Length(seconds)
IVR_VOICEMAILWELCOME	voicemailwelcome	Default Audio for Voice Mail Access

4.4 Softswitch Recorded CDR File In Txt

When the softswitch's parameter "SS_CDRRECORDTOFILE" is set to On, the softswitch will record cdr into txt file in the directory cdr beneath the installation directory.

The format of the cdr file name will be YYYYMMDDHH.txt(YYYY-year, MM-month, DD-day, HH-hour).

The softswitch will generate one cdr file per hour, as 2013103112.txt will record the the cdrs that end between 2018-12-20 12:00 and 2018-12-20 13:00.

Each line represents one cdr in the txt file. The format is tabled as below:

```
callerE164 |calleeE164 |startTime |stopTime |holdTime |endReason |endDirection
|callerGatewayId |calleeGatewayId |callerIp |calleeIp |callerAccessE164 |calleeAccessE164
|callerToGatewayE164 |calleeToGatewayE164 |calleeBilling |billingMode |callerPdd
|calleePdd
```

segment	description
callerE164	The caller id
calleeE164	The callee id
startTime	Begin time, as 2018-12-20 11:20:18
stopTime	End time, as 2018-12-20 16:34:09
holdTime	Call durarion (in milliseconds)
endReason	End reason
endDirection	Hangup side (0-caller, 1-callee, 2-server)
callerGatewayId	Calling gateway
calleeGatewayId	Called gateway
callerIp	Caller ip
calleeIp	Callee ip
callerAccessE164	Incoming caller
calleeAccessE164	Incoming callee
callerOutE164	Outbound caller

calleeOutE164	Outbound callee
calleeBilling	Billing method (0-By caller, 1-By callee)
billingMode	Charge mode (-1-bobilling, 0-phone number, 1-gateway ID, 3-phone card)
callerPdd	time elapsed from call received to call connected
calleePdd	time elapsed from call sent to routing response

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4.5 Call End Reason

4.5.1 Server Side

Terminations caused by the server are all defined by VOS3000. For any questions related to this kind of causes, please consult your system supplier.

Following are the causes of this kind:

- Response timeout: the called did not answer the phone before the timeout limit was reached. The timeout limit can be set either by the “Alerting” signal of Routing Gateway (when the call is processed through Routing Gateway) or by the “SS_TIMEOUT_PHONE_HANGUP” parameter in the Softswitch system (when the call is directed to a phone)
- Protocol limit: the server cannot process this type of protocol
- Connection timeout: No response to the SIP message was received after specified number of trials. The maximal number of trials can be specified by the “SS_SIP_RESEND_INTERVAL” and “SS_SIP_SEND_RETRY” parameters in the Softswitch system.
- Busy: the number being called is busy.
- Account locked: the account is disabled. It may also due to the lockdown of its agent account.
- Session timeout: the caller or the called supports the SIP Timer protocol, yet it has not received the updating signal within a time limit; or, neither does the caller nor the called support the SIP Timer protocol, yet the session time exceeded the value specified by the system parameter
- SS_SIP_NO_TIMER_REINVITE_INTERVAL”.
- Caller’s number restricted: the length of the caller’s number exceeds the length specified by the system parameter “SS_CALLERALLOWLENGTH”.
- Called number restricted: the length of the called number exceeds the length specified by the system parameter “SS_CALLERALLOWLENGTH”.
- Proceeding timeout: No response is received from the server within the time limit. The time limit can be specified by the “Setup” and “Callproceeding” parameters in the routing gateway.
- Caller’s number prefix restricted: the mapping gateway does not accept the caller’s number prefix.
- Unregistered: the terminal is not register, and is not allowed to make the call.
- Called number locked: the called is currently locked.
- Called number prefix restricted: the gateway for the caller or the called gateway does not accept the called number prefix.
- Caller locked: the caller is currently locked.
- Connection establishment timeout: the connection is not established within the timeout limit set by the mapping gateway. The time limit can be changed by setting the proceeding timeout parameter in the mapping gateway.
- Account expired: the account is expired.
- Connection limit exceeded: the maximum number of outgoing calls is reached. The maximum number can be specified in the system.

- Forcible hang-up: the server disconnected the session, usually because the client chose to end the session in their user interface.
- Account disabled: the account is currently disabled. Please check the status of the account it belongs to.
- The called not online: There is no appropriate device to accept this call. For example, there is no matching routing gateway.
- No-answer forwarding by the caller: the caller has set the no-answer forwarding
- Timed forwarding: the call matches the timed forwarding criteria specified in the phone management settings.
- On-busy forwarding: the call matches the on-busy forwarding criteria specified in the phone management settings.
- No-answer forwarding by the called: the call matches the no-answer forwarding criteria specified in the phone management settings.
- Forwarding loop: Due to the wrong configuration of users, the forwarding route has loops.
- Call forwarding by the called: the call matches the call forwarding criteria specified in the phone management settings.
- Do-not-disturb from the called: the called is in the do-not-disturb status
- Session closed by the called: the called did not send the hang-up signal, but disconnected the TCP connection
- Session closed by the caller: the caller did not send the hang-up signal, but disconnected the TCP connection
- Illegal call: the call comes from an unauthorized IP address and the caller's number is not registered in the system.
- No matching rate: There is no rate that matches this call.
- No matching account: There is no account to bill this call.
- Insufficient balance: the account has insufficient balance
- Call restriction: the call is prevented by restrictions (such as <International call>) set by the phone or the gateway.
- Hang-up by the called: the hang-up signal comes from the called.
- Hang-up by the caller: the hang-up signal comes from the caller.

4.5.2 Client Side

4.5.2.1 H323 Device

UnknownCauseIE
UnallocatedNumber
NoRouteToNetwork
NoRouteToDestination
SendSpecialTone
MisdialledTrunkPrefix
ChannelUnacceptable
CallAwarded

Preemption
PreemptionCircuitReserved
NormalCallClearing
UserBusy
NoResponse
NoAnswer
SubscriberAbsent
CallRejected
NumberChanged
Redirection
ExchangeRoutingError
NonSelectedUserClearing
DestinationOutOfOrder
InvalidNumberFormat
FacilityRejected
StatusEnquiryResponse
NormalUnspecified
NoCircuitChannelAvailable
CallQueued
NetworkOutOfOrder
FrameModeOOS
FrameModeOperational
TemporaryFailure
Congestion
AccessInformationDiscarded
RequestedCircuitNotAvailable
PrecedenceCallBlocked
ResourceUnavailable
QoSNotAvailable
RequestedFacilityNotSubscribed
OutgoingCallsBarred
OutgoingCallsBarredInCUG
IncomingCallsBarred

IncomingCallsBarredInCUG
BearerCapNotAuthorised
BearerCapNotPresentlyAvailable
InconsistentOutgoingIE
ServiceOptionNotAvailable
BearerCapNotImplemented
ChannelTypeNotImplemented
RequestedFacilityNotImplemented
OnlyRestrictedDigitalBearerCapAvailable
ServiceOrOptionNotImplemented
InvalidCallReference
IdentifiedChannelNonExistent
CallIdentifyNotSuspendedCall
CallIdentifyInUse
NoCallSuspended
ClearedRequestedCallIdentity
UserNotInCUG
IncompatibleDestination
NonexistentCUG
InvalidTransitNetwork
InvalidMessageUnspecified
MandatoryIEMissing
MessageTypeNonexistent
MessageNotCompatible
IENonExistantOrNotImplemented
InvalidIEContents
MessageNotCompatibleWithCallState
TimerExpiry
ParameterNonexistent
UnrecognisedParamaterDiscarded
ProtocolErrorUnspecified
InterworkingUnspecified
ErrorInCauseIE

4.5.2.2 SIP Device

Multiple Choices
Moved Permanently
Moved Temporarily
Use Proxy
Alternative Service
Bad Request
Unauthorized
Payment Required
Forbidden
Not Found
Method not Allowed
Not Acceptable
Proxy authentication Required
Request Timeout
Gone
Request Entity Too Large
Request-URI Too Long
Unsupported Media Type
Unsupported URI Scheme
Bad Extension
Extension Required
Session Interval Too Small
Interval Too Brief
Temporarily Unavailable
Call/Transaction Does not Exist
Loop Detected
Too Many Hops
Address Incomplete
Ambiguous
Busy Here
Request Terminated
Not Acceptable Here

Request Pending
Server Internal Error
Not Implemented
Bad Gateway
Service Unavailable
Server Time-out
Version not Supported
Message Too Large
Busy Everywhere
Decline
Does not Exist Anywhere
Not Acceptable

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5 Maintenance

About This Chapter

This chapter describes the maintenance of VOS3000 solutions.



CAUTION

If one server deployed multiple services, should start and stop in the following order.

5.1 All Services Restart

```
servermonitord VOS3000 restart  
all
```

5.2 All Services Stop

```
servermonitord VOS3000 stop all
```

5.3 All Services Start

```
servermonitord VOS3000 start  
all
```

5.4 Stop Process

Root Permission:

```
systemctl stop vos3000d  
systemctl stop mgcd
```

```
systemctl stop empd
systemctl stop callserviced
systemctl stop mbx3000d
systemctl stop webdatad
systemctl stop webserverd
systemctl stop audioplayerd
systemctl stop serverrmonitord
```

Non Root Permission:

```
VOS3000d stop
mgcd stop
empd stop
callserviced stop
mbx3000d stop
webdatad stop
webserverd stop
audioplayerd stop
servermonitord stop
```

5.5 Start Process

Root Permission:

```
systemctl start VOS3000d
systemctl start mgcd
systemctl start empd
systemctl start callserviced
systemctl start mbx3000d
systemctl start webdatad
systemctl start webserverd
systemctl start audioplayerd
systemctl start servermonitord
```

Non Root Permission:

```
VOS3000d resume
mgcd resume
empd resume
callserviced resume
mbx3000d resume
webdatad resume
webserverd resume
```

```
audioplayerd resume  
servermonitord resume
```

5.6 Master Server Lock

```
/etc/init.d/masterslaved lock
```

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