Action URL and Active URI Specifications

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Contents

1 Introduction

1.1 Overview

Action URL is used by a telephone set to report the current status to the remote console.
Active URI is used by a remote console to control various operations of a telephone set.
Action URL and Active URI are mainly used in Computer Telephony Integration (CTI) scenarios. A telephone set reports its own status to a PC and the PC controls the operations of the telephone set. Typical application scenarios include the call center. An attendant operates the telephone set on the PC by using the CTI software.

2 Action URL

2.1 Introduction

When the status of a telephone set changes, the telephone set sends an HTTP Get request to the remote console to notify this event. The console can perform corresponding operations on the telephone set based on the status change.

2.2 Protocol Description

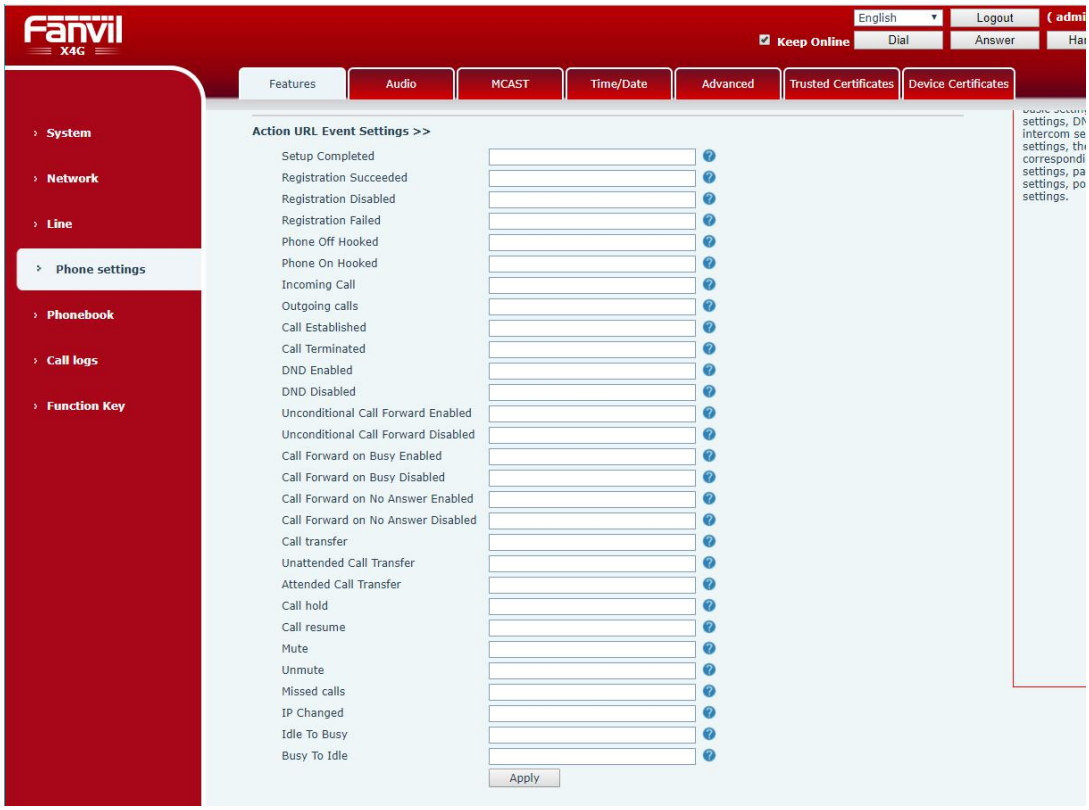
The HTTP URL format of Action URL is defined by the HTTP server of the console. The telephone set is responsible only for initiating an HTTP Get request upon a status change. Generally, the HTTP URL format of Action URL is as follows: http://192.168.1.100 /newcall.xml?num=$call_id

- 192.168.1.100 is the IP address of the remote console.
- newcall.xml? indicates the method of handling different statuses. The method is defined by the HTTP server of the remote console.
- $call_id indicates an internal variable of the telephone set. Before an HTTP Get request is initiated, the system automatically replaces this variable with the actual current value of the system. The internal variable begins with a dollar sign ($).

2.3 Action URL Configuration

Log in to the management webpage of the telephone set and choose Phone > Feature > Action URL Settings. Enter the corresponding URL in the text box of each event. For example, enter http://192.168.1.100 /newcall.xml?num=$call_id after the Incoming Call event.

When a new incoming call from number 1234 is received after configuration, the telephone set initiates HTTP Get http://192.168.1.100 /newcall.xml?num=1 (sequence number of the call).



2.4 Event List

Event	Description
Setup Completed	The telephone set is started successfully.
Registration Succeeded	An account is registered successfully.
Registration Disabled	Account registration is canceled.
Registration Failed	Account registration fails.
Phone Off Hooked	The telephone set is hooked off.
Phone On Hooked	The telephone set is hooked on.
Incoming call	A new incoming call is received.
Outgoing call	An outgoing call is made.
Call established	A call is connected.
Call terminated	A call is terminated.
DND Enabled	Do Not Disturb (DND) is enabled.
DND Disabled	DND is disabled.
Unconditional Call Forward Enabled	Unconditional call forwarding is enabled.
Unconditional Call Forward Disabled	Unconditional call forwarding is disabled.
Call Forward on Busy Enabled	Call forwarding on busy is enabled.
Call Forward on Busy Disabled	Call forwarding on busy is disabled.
Call Forward on No Answer Enabled	Call forwarding on no answer is enabled.
Call Forward on No Answer Disabled	Call forwarding on no answer is disabled.

Call transfer	Call transfer.
Unattended Call Transfer	Unattended call transfer.
Attended Call Transfer	Attended cal transfer.
Call hold	Call hold.
Call resume	Call hold is canceled.
Mute	A call is muted.
Unmute	A call is unmuted.
Missed calls	Missed calls are listed.
IP Changed	The IP address of the telephone set is changed.
Idle To Busy	The telephone set switches from the standby screen to other screens.
Busy To Idle	The telephone set switches from other screens to the standby screen.
MWI	Message.
SMS	SMS message.
Start reboot	The telephone set is restarted.
Close The Door	Door lock status open and close
Close The Door	Door lock status open and close
Tamper	Trigger anti-demolition alarm
Door Sensor 1	Door magnetic detection alarm use
Door Sensor 2	Door magnetic detection alarm use
Web API Auth Changed	Web API Auth Changed
Received Sip Message	Received Sip Message
Output1	Output1 triggers an alarm, the corresponding interface changes potential, and an alarm bell rings
Output2	Output1 triggers an alarm, the corresponding interface changes potential, and an alarm bell rings

Note: blue font is industry specific

2.5 Variable List

Variable	Description
$mac	Device MAC address.
$ip	Current available IP address.
$model	Model of the telephone set.
$firmware	Software version.
$active_uri	Session Initiation Protocol (SIP) URI of the current active account, which is valid in incoming calls, outgoing calls, and conversations
$active_user	User account of the SIP URI of the current active account, which is valid in incoming calls, outgoing calls, and conversations
$active_host	Server of the SIP URI of the current active account, which is valid in incoming calls, outgoing calls, and conversations
$local	X3/4 series: local SIP URI (valid in outgoing calls) Local phone number (valid in incoming calls before they are answered) X6: local SIP URI (valid in incoming calls, outgoing calls, and conversations)
$remote	X3/4 series: remote SIP URI (valid in incoming calls) Remote phone number (valid in outgoing calls before they are answered) X6: remote SIP URI (valid in incoming calls, outgoing calls, and conversations)
$display_local	Local display name (phone number displayed if no display name is set) (valid in incoming and outgoing calls)
$display_remote	X3/4 series: remote display name (phone number displayed if no display name is set) (valid in incoming calls) X6: remote display name (phone number displayed if no display name is set) (valid in incoming and outgoing calls)
$call_id	Call ID (valid in incoming calls, outgoing calls, and conversations)
$duration	Call duration (valid when a conversation ends)
$date_time	Acquisition time
$memory_free	Memory
$flash_free	Flash memory (not implemented yet)
$line	Call line (valid in incoming calls, outgoing calls, conversations, and registration)
$local_user	Local users in a conversation (valid in incoming calls, outgoing calls,

	and conversations)
$local_server	Server used in a SIP call (valid in incoming calls, outgoing calls, and conversations)
$local_domain	Domain of a SIP cal (valid in incoming calls, outgoing calls, and conversations)
$local_number	Local phone number during a call (valid in incoming calls, outgoing calls, and conversations)
$local_displayname	Display name of the local phone number during a call (valid in incoming calls, outgoing calls, and conversations)
$remote_number	Remote phone number during a call (valid in incoming calls, outgoing calls, conversations, and unanswered incoming calls)
$remote_displayname	Display name of the remote phone number during a call (valid in incoming calls, outgoing calls, and conversations)

Note: The variables highlighted in green are valid only in X6.

Variable description:

1) For a variable valid only in incoming calls, this variable is replaced with the corresponding information only if it is set in Incoming call options.
2) For a variable valid only in outgoing calls, this variable is replaced with the corresponding information only if it is set in Outgoing call options.
3) For a variable valid only in conversations, this variable is replaced with the corresponding information only if it is set in conversation related options such as Call established, Call terminated, Transfer call, Blind transfer call, Attended transfer call, Hold, Unhold, Mute, and Unmute.

3 Active URI

3.1 Introduction

The remote console initiates an HTTP Get request, the HTTP server embedded into the telephone set parses the instruction and responds to the request to remotely control the telephone set.

3.2 Set Active URI Limit IP

Set the IP that allows remote operation of the Phone through the web page >Phone Settings> Features> Active URI Source IP.

Restrict Active URI Source IP parameter states:

Parameter name	Restrict Active URI Source IP
Parameter description	1. Push the active URI command to HTTP protocol: 1) When the IP list is not empty, only the filled IP can send HTTP active URI command to the phone; 2) When the IP list is empty, the phone accepts any HTTP active URI commands pushed by IP 2. The active URI command pushed by SIP notify: 1) Notify from registered SIP server Do not check the IP restriction list, accept by default; 2) IP is empty. Any IP other than SIP registration server sends active URI through SIP notify, and the phone does not respond IP is not empty. Only the listed IP and sip registration server IP can operate the phone directly. When the IP in the list sends the SIP notify message of event: active-uri to the phone, the phone will respond directly without pop-up prompt
Effective value	IP address
Default value	empty

3.3 Instruction List

Access dial	
key=SPEAKER | key= F_HANDFREE | key= F_HANDSFREE	Hands free key, turn on and off handsfree
F_HEADSET（or key=HEADSET）	Headset key
key=F_ Prefix: prefix number | key = f_ Prefix; prefix number	prefix
Answer the phone	
key=OK | key=ENTER | key=F_OK	Confirm key, you can answer the phone; pick up the phone; send the number; enter the menu under standby
key=F_ACCEPT	Answer key
CALL	
key=SPEAKER;0000;ENTER	Call with handsfree carrying number 0000
key=0000;ENTER	Call with number 0000 in standby mode
key=F_LOR	call back number
key=F_HEADSET;000;ENTER | key=HEADSET;000;ENTER	Headset mode carries the number 000 to initiate a call
key=RD | key=F_REDIAL	Redial the number and press the redial key
F_SEND	When there is a number on the dial, sending out the number is the same as pressing the daily key
Hang up or reject	
key=RELEASE | key=F_RELEASE	It's the same as pressing release Hang up, refuse to answer, quit dialing, etc Exit an application interface, etc
key=F_CANCEL | key=X	It's the same as pressing cancel
key=F_REJECT	It's the same as pressing the reject key
transfer	
key=F_TRANSFER | key=F_B_TRANSFER | key=F_A_TRANSFER	It works the same as pressing the transfer key It can realize blind transfer, present transfer and semi attendance transfer
key=F_TRANSFER;0000;OK	Call 0000
key=F_TRANSFER;000;F_A_TRANSFER	Blind to 000
key=F_DIVERT	Call forwarding
conference	

key=F_CONFERENCE	The same effect as pressing the conference key will enter the conference dialing interface
key=F_JOIN	Join a three-way call just like pressing dskey join
Advanced call function	
key=DND	On / off dnd
key=DNDON	On DND
key=DNDOFF	Off DND
F_PICKUP	It is the same as pressing dsskey pickup to answer the call and call normally
key=F_PARK	As with pressing dskey Park, the call is resident
key=F_AUTOREDIAL	As with pressing dskey automatic, automatic dialing (which takes effect when dialing) is only valid for X4 series
key=F_UNAUTOREDIAL	The same as pressing dskey unautoredial, cancels the automatic dialing (it takes effect when dialing), and only works for X4 series
key=1234	In the call state, input DTMF 1234 continuously
key=F_HOLD	Hold / release hold
key=F_0-9/*/POUND	Enter a single DTMF number (including numeric keys and *)
key=* | key=F_* | key=F_STAR	*Key
key=# | key=POUND | key=F_ POUND	# Key
F_REC	Recording during a call
key=F_FLASH	Switching calls / answering calls during calls
key=F_GROUPLISTEN	Group listening
key=exit	Exit, only supported by X6 series
key=clear	Dialing under clear input number, only X6 series support
key=split	Release meeting, only X6 series support
key=prev_line/ key=next_line	Switch line line, only supported by X6 series
key=prev_call/ key=next_call	Switching calls, only supported by X6 series
BroadSoft related	
key=F_DISPOSITION	BroadSoft call center application mark call record type

key=F_ESCALATE	Broadsoft CallCenter In the application, the function of "one key to join the manager" is realized
key=F_TRACE	Broadsoft CallCenter The function used to inform the server to record a call in the application
key=F_PRIVATEHOLD	In BroadSoft shared call appearance, the function of holding the other party but not sharing the information is relative to the public hold key
modulation	
key=VOLUME_UP | key=F_VOLUME_UP	It works the same as pressing the volume down button The volume can be turned up in talking, conf, idle and other states
key=VOLUME_DOWN | key=F_VOLUME_DOWN	Reduce the volume
key=F_MUTE|MUTE	Silence
Dsskey	
key=L1-L6 | key=F_L1-F_L6	Line key, only for X4 series
key=F_D1-F_D12	Dsskey, available for X4 series only
key=DSS1-DSS12	Valid for X6 series only
key=F1-F4/ key=F_F1-F_F4	Softkey 1-4
Enter the app	
key=F_AGENT	Enter the agent configuration interface
key=MSG | key=F_MWI	Access to voice mail
key=F_MEMO	Enter memo
key= F_PBOOK	Access to the phone book
key=F_LOCALCONTACTS	Local phone book
key=F_SERVICE	Access to Internet phone book
key=F_SMS	Enter the SMS interface
key=F_LOCK	Enter the keyboard lock configuration interface
key=F_SDTMF	Enter the hide DTMF configuration interface
key=F_CFWD	Enter call forwarding settings
Modify configuration / restart / restore factory settings	
key=F_HOTDESKING	Clear SIP configuration

key=Reboot | key=F_REBOOT	restart
key=AutoP	Restart autoprovision detection
line=x;displayname=xxxxx	Modify the display name of X line
key=Reset	Restore factory settings
Navigation key / delete key / menu key	
key=UP/key=F_UP	On the navigation key
key=DOWN/F_DOWN	Navigation key down
key=LEFT/F_LEFT	Navigation key left
key=RIGHT/F_RIGHT	Navigation key right
key=OK	Navigation key OK
key= F_DELETE | key=DELETE	Delete key
key=menu	Menu, only supported by X6 series
Open door (cooperate with access control products)	
key=F_LOCK&code=openCode	Realize the function of remote door opening

Note: blue font is industry specific

3.4 Protocol Description

The format of the HTTP URL of Active URI is as follows: http://192.168.1.190/cgi-bin/ConfigManApp.com?key=OK"

- 192.168.1.190 is the IP address of the telephone set.
- "/cgi-bin/ConfigManApp.com" is the fixed format of Active URI. The telephone set parses it and considers it as the control instruction of Active URI.
- "?key=OK" is the instruction type indicating the action to be performed by the telephone set. Generally, this instruction is a key event.

In some environments, HTTP authentication is enabled by default for telephone sets at delivery to ensure security. The HTTP URL format is as follows: http://admin:admin@192.168.1.190/cgi-bin/ConfigManApp.com?key=OK

"admin:admin" indicates the default authentication user name and password of the HTTP server embedded into the telephone set. The user name and password are consistent with those used for logging in to the management webpage.

3.5 Sending a Group of Numbers

The preceding Active URI instruction supports initiating only one operation event with the telephone set at a time. For example, the instruction supports entering only one key at a time

when a user enters a number, making the operation complex. Active URI supports sending a group of numbers in the URL.

- Initiating a call with a number
  http://admin:admin@192.168.1.190/cgi-bin/ConfigManApp.com?key=SPEAKER;000;ENTER
  When the telephone set receives this instruction in the standby state, it enters the hands-free off-hook state and automatically initiates a call to number 000.
- Blind transfer with a number
  http://admin:admin@192.168.1.190/cgi-bin/ConfigManApp.com?key=F_TRANSFER;000;F_TRANSFER
  When the telephone set receives this instruction during a call, it automatically transfers the current call to number 000.

3.6 Application Examples

For example, the call center realizes the following scenarios through the active URI command: answering a call, answering a second call, switching calls between two calls, transferring the second call to the present, etc

1) Answer a call: when you receive a call, use the http: / / phone IP / CGI bin/ ConfigManApp.com?key=OK Answer the call;

2) Answer the second call: after receiving the second call, use http: / / phone IP / CGI bin/ ConfigManApp.com?key=OK Answer the second call; the first call is automatically held.
3) Switching between two calls: there are two ways to switch calls: one is to use F_ Flash command, one is to use prev_ call/next_ Call and F_ The combination command of hold has the following differences:

- It is assumed that the telephone has a, B and C 3-way calls, and the current call path is a, and B / C is in hold state. Using http: / / phone / CGI bin/ ConfigManApp.com?key=F_ Flash switches the current channel. After inputting the command, B switches to call status, and a / C is hold state
- It is assumed that the telephone has a, B and C 3-way calls, and the current call path is a, and B / C is in hold state. Using http: / / phone IP / CGI bin/ ConfigManApp.com?key=prev_ Call (or next_ Call) to call B, and then through IP / CGI bin/ ConfigManApp.com?key=F_ Hold will switch bresume to call status.
  The corresponding scheme can be selected according to the actual scene

Transfer the second way call: use the method in step 3) to switch the call to the second channel (at this time, the second way is hold or the call status is OK, which does not affect the transfer), and use http: / / phone IP / CGI bin/ ConfigManApp.com?key=F_ Transfer; 0000; OK command

will transfer the call attendance to 0000 (or other number), after 0000 answers, use http: / / phone IP / CGI bin/ ConfigManApp.com?key=OK Order completion of attendance transfer.

3.7 SIP Notify Push Active URI Command

1. Notify format

When a sip notify message with event: active URI is received by the phone, the instruction with the format of key = XXX in the message body will be executed by the caller.

Notice format reference:

```
NOTIFY sip:3583@10.2.40.10:5062 SIP/2.0
Via: SIP/2.0/UDP 10.2.40.27:5063;branch=z9hG4bK4163876675
From: <sip:3586@10.2.1.48 > ;tag=2900480538
To: "3583" <sip:3583@10.2.1.48 > ;tag=490600926
Call-ID: 2923387519@10.2.40.10
CSeq: 4 NOTIFY
Contact: <sip:3586@10.2.40.27:5063 >
Max-Forwards: 70
User-Agent: Fanvil X6 1.12.5
Event: ACTIVE-URI
Content-Type: message/sipfrag
Content-Length: 6

key=OK
```

2. IP limit of push SIP notify command: refer to 3.2 Active URI Limit IP。

3. Uri Directive: Reference 3.3 Instruction List。