SWIPPA Quick Start Manual

History:

Time	Content	Author	Version
2005-06-14	1st version	Jianming. Feng Senior R&D Manager.	1.0
2010-02-22	2nd version	Jianming. Feng Cindy. Shen	1.1
2010-03-05	3rd version	Jianming. Feng	2.0

Content

1 Introduction

1.1 Overview of SWIPPA

SWIPPA™ is a VoIP recording component, with high customizability and compatibility in any special situation. SWIPPA™ module is designed around client/server architecture, in which the client is the application requesting the recording service, and the server providing the service is the recording module. The client/server software architecture is a versatile, message-based and modular infrastructure that is intended to improve flexibility and interoperability as compared to centralized and time sharing processing. A single machine can play both client and server roles, depending on the software configuration, which means both application and the recording modules can be loaded into one machine. Moreover, the client can be served by multiple recording servers.

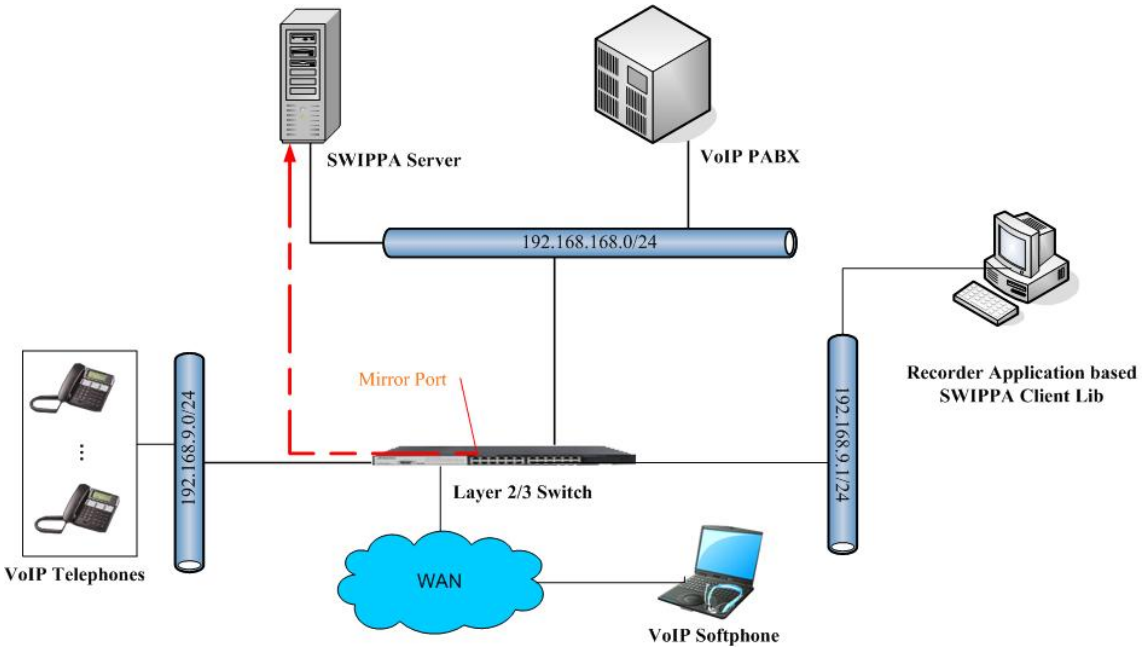
SWIPPA consists of SWIPPA runtime library and SWIPPA SDK.

- SWIPPA runtime library is to direct VoIP protocol implementation and provide recording control interface.
- SWIPPA SDK provides recording application interface and related documents.

1.2 Deploy SWIPPA

The PC server (in which SWIPPA Server is loaded), should have been installed with two network cards. One network card is used to receive IP package from the VOIP phone set and connect with mirror port of the Switch, and can be called “recording network card”. Another card is connected to common ports of the Switch for communication purpose, and can be defined as “communication network card”. All IP packages from the VOIP phone can be tapped and then copied to the mirror port, so that SWIPPA Server can analyze these IP packages for recording purpose.

Recorder Application and SWIPPA Server developed from SWIPPA Client Lib, can be deployed on single PC or on multiple PC servers. Multi-to-multi connection is used between Recorder Application and SWIPPA Server.

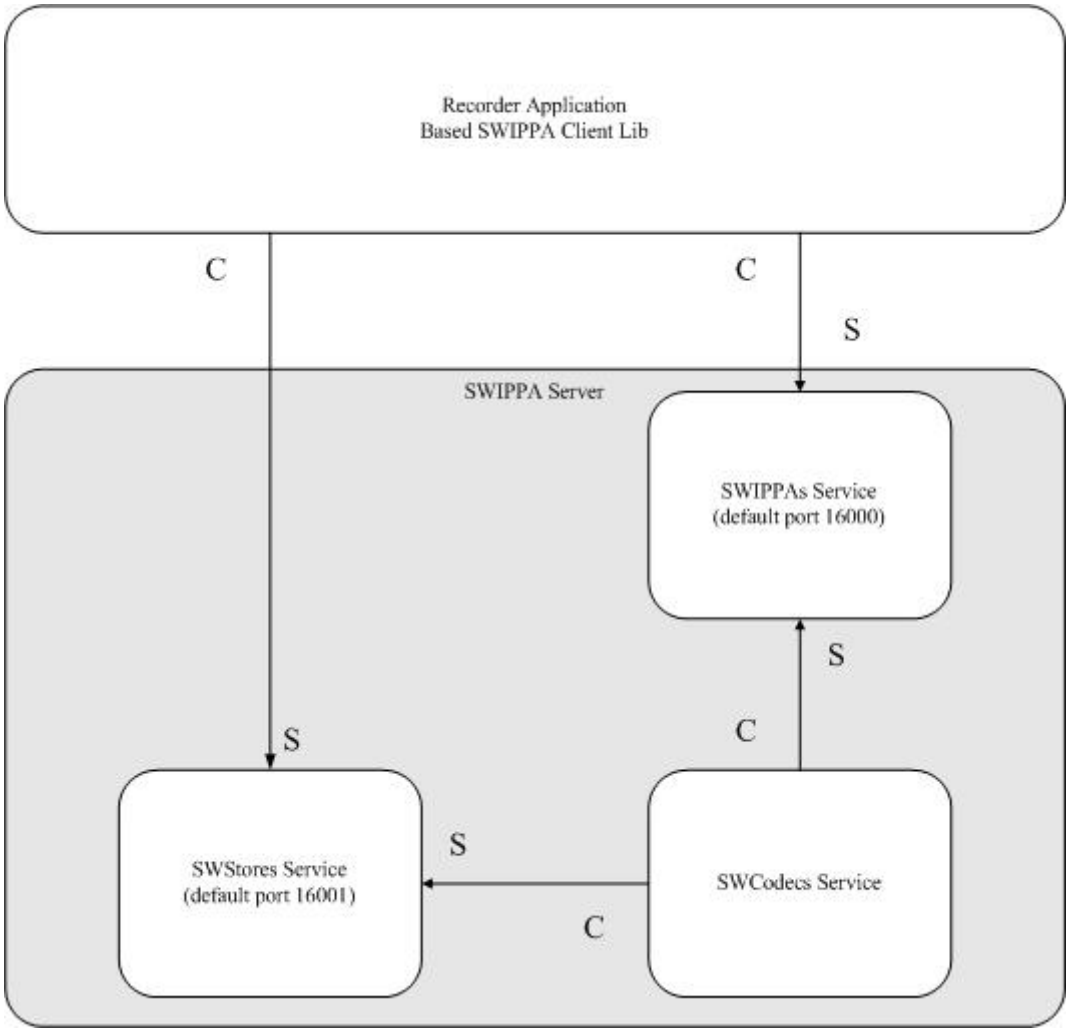


1.3 SWIPPA Runtime Library

SWIPPA runtime library includes SWIPPA Server runtime library and SWIPPA Client runtime library.

- SWIPPA Server runtime library is for parsing IP packages for recording purpose.
- SWIPPA Client runtime library provides recording control interface.

SWIPPA Server offers three types of services: SWIPPAs Service, SWCodecs Service and SWStores Service.



Modules description:

Number	Module	Functional description
1	SWIPPA Client Lib	Recorder Application communicates with SWIPPA Server via SWIPPA Client Lib interface. TCP/IP communication is used between SWIPPA Client Lib and SWIPPA Server
2	SWIPPAs Service	This module is to capture and parse VOIP package
3	SWCodecs Service	This module is to set recording CODECs and mixer
4	SWStores Service	This module is to store recorded file and provide voice stream for live monitoring purpose

1.4 SWIPPA Release Package

You can download SWIPPA Release package from Synway website or ask for SWIPPA release package CD, both of which provides SWIPPA installation package, WIPPA SDK package and documents. http://www.synway.net/Products/index_xx.aspx?id=72.

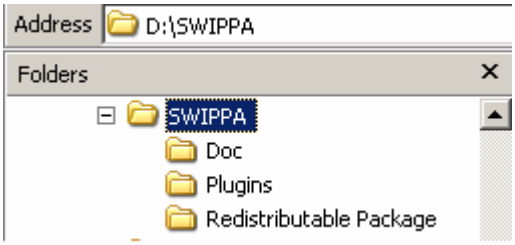
1.4.1 SWIPPA Installation Package

SWIPPA installation package include SWIPPA Server runtime library and SWIPPA Client

runtime library.

1. Installation directory for SWIPPA Server

Installation directory of SWIPPA Server is shown in the figure below.



- Doc: to store documents.
- Plugins: to store plugins DLLs. A plugin file is provided for each PBX.
- Redistributable Package: to store SWIPPA Client runtime library.

2. SWIPPA Client runtime library

SWIPPA Client runtime library can be found in the directory of Redistributable package under SWIPPA installation path. All files listed in the table below shall be copied to a folder named as Redistributable package in the recording application.

Number	File name	Description
1	SWIPPA32c.dll	SWIPPA Client Lib dynamic link library
2	SWStore32c.dll	Voice stream transmission dynamic link library
3	SWGCI32c.dll	TCP/IP communication dynamic link library

1.4.2 Documents

Number	Document name	Description
1	SWIPPA Quick Start Manual	For software development engineers, software deployment engineers
2	SWIPPA SDK User Manual	For software development engineers

1.4.3 SWIPPA SDK package

For how to use SWIPPA SDK package, please refer to the document “SWIPPA SDK User Manual”.

1.5 SWIPPA Packaging List

Number	File name	Description
1	SWIPPA release package	Where to get: you can download the package from the synway website or ask sales representatives for the package CD
2	USB dongle	Each USB dongle has a unique code Where to get: purchase it from Synway sales representatives

Number	File name	Description
3	Authorization code	Authorization code is to determine who can use the software and how long the software can be used Where to get: purchase from Synway sales representatives

2 Installation

1. Deploy IP network environment

To tap IP package into the mirror port, a Switch with mirror ports is required to mirror all IP packages through IP phone. The recording network card in SWIPPA Server is connected to mirror port of the Switch. And the communication network card is connected to common ports of the Switch.

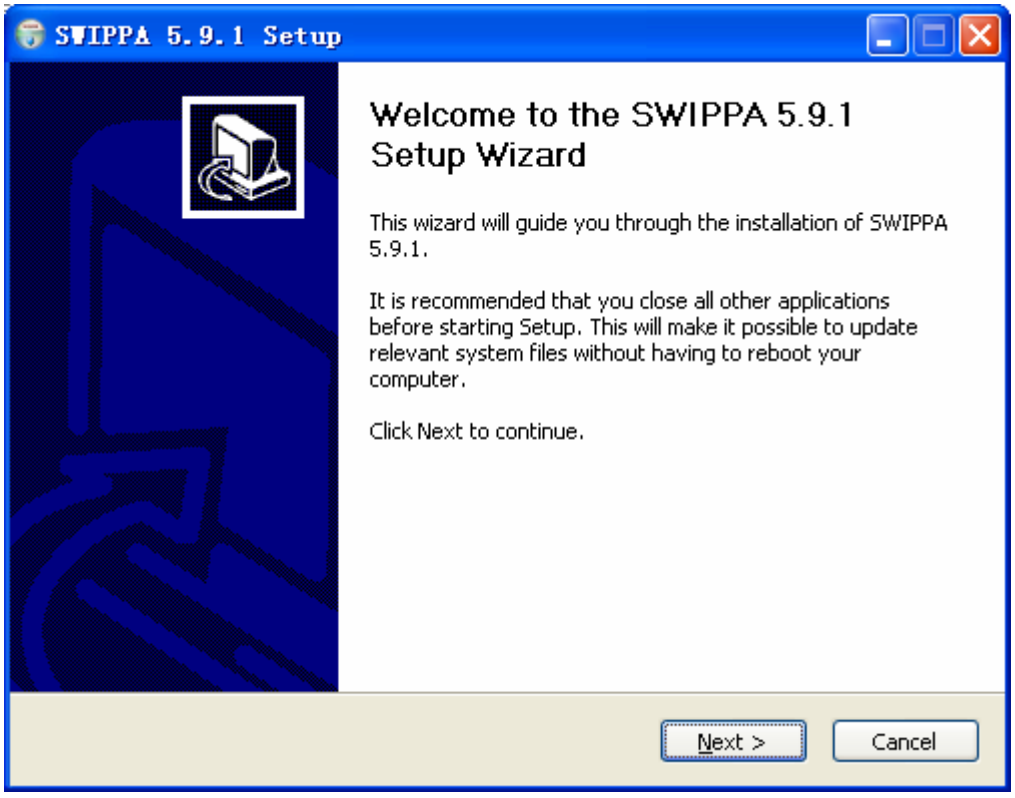
2. Install USB dongle

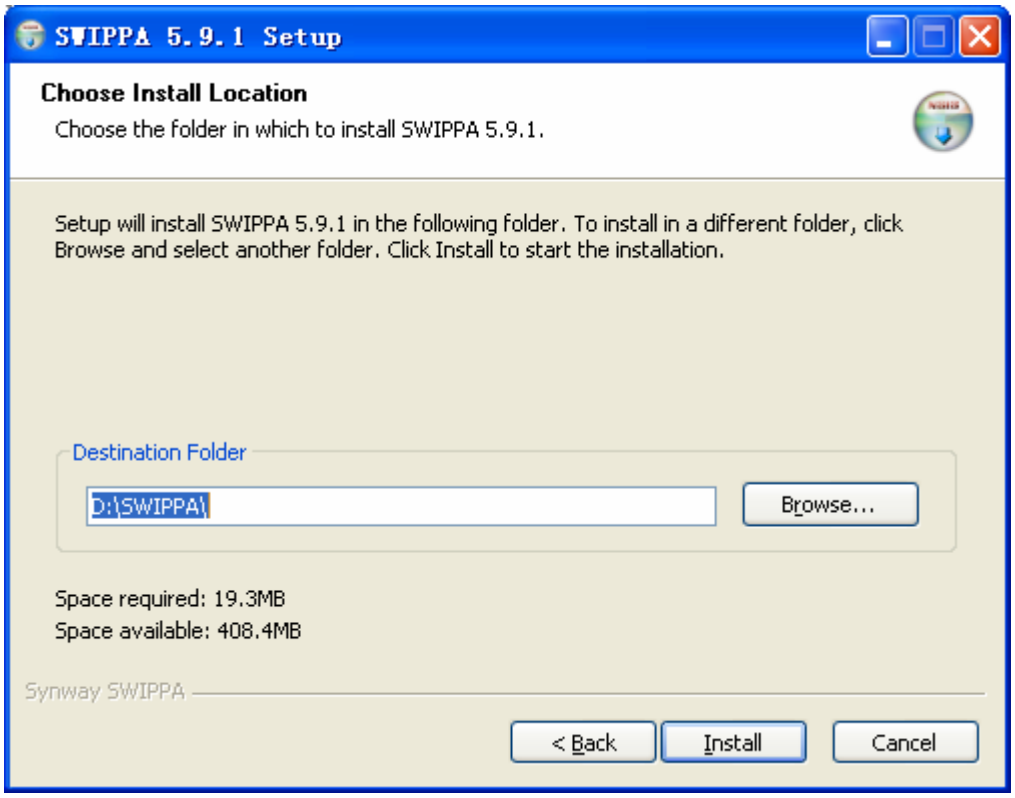
USB dongle must be purchased from Synway or Synway authorized sales partners. USB dongle should plug into the server in which SWIPPA Server is installed. DO NOT plug or loosen the dongle during system run time.

3. Install SWIPPA Server

Executable file SWIPPA5.9.1-Setup.exe is the setup file for SWIPPA Server. Thereof, 5.9.1 indicates SWIPPA version.

Execute SWIPPA5.9.1-Setup.exe, the Window like in the below figure pops up, and complete installation of SSWIPPA Server according to the installation wizard.

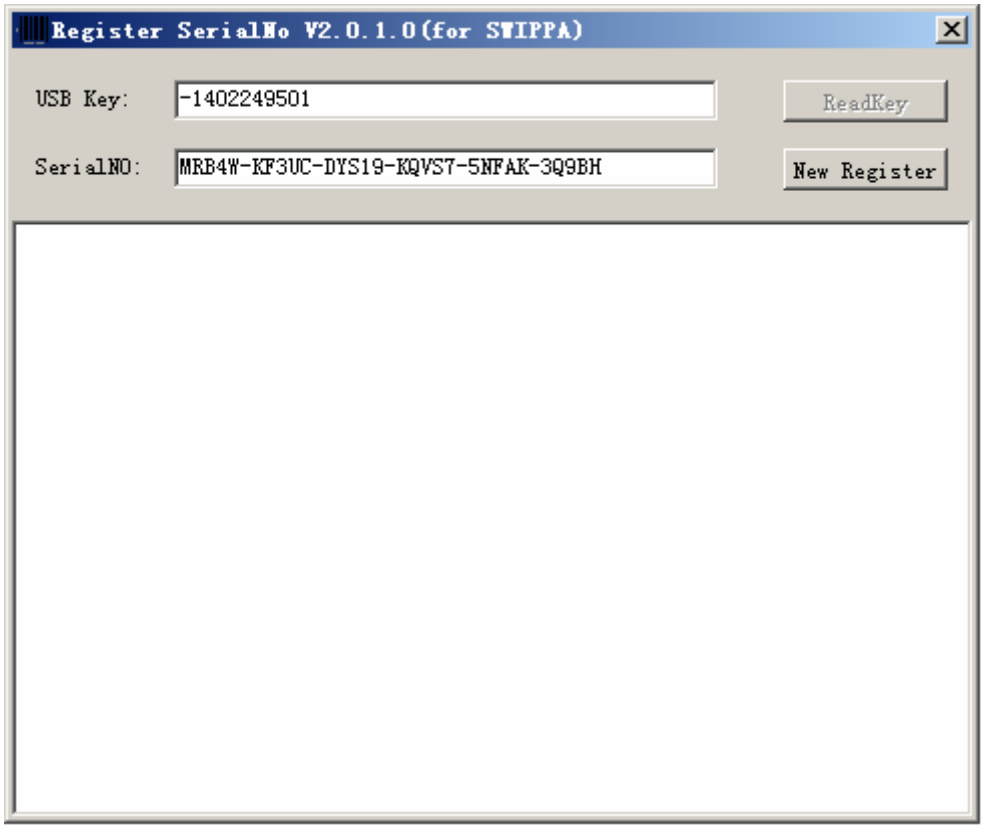




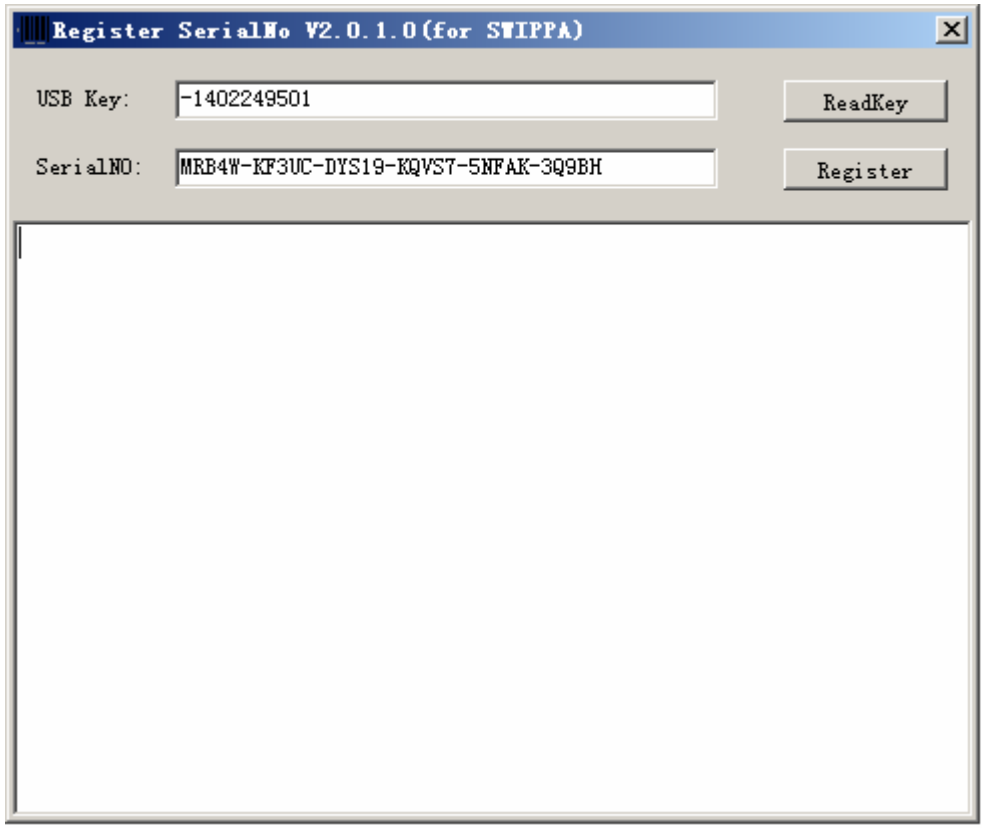
As shown in figure above, click button <Browse> to set installation path.

3 Enter authorization code

Select menu [Start/Synway SWIPPA/SWIPPA RegSerialNo], and the Window pops up as shown below.

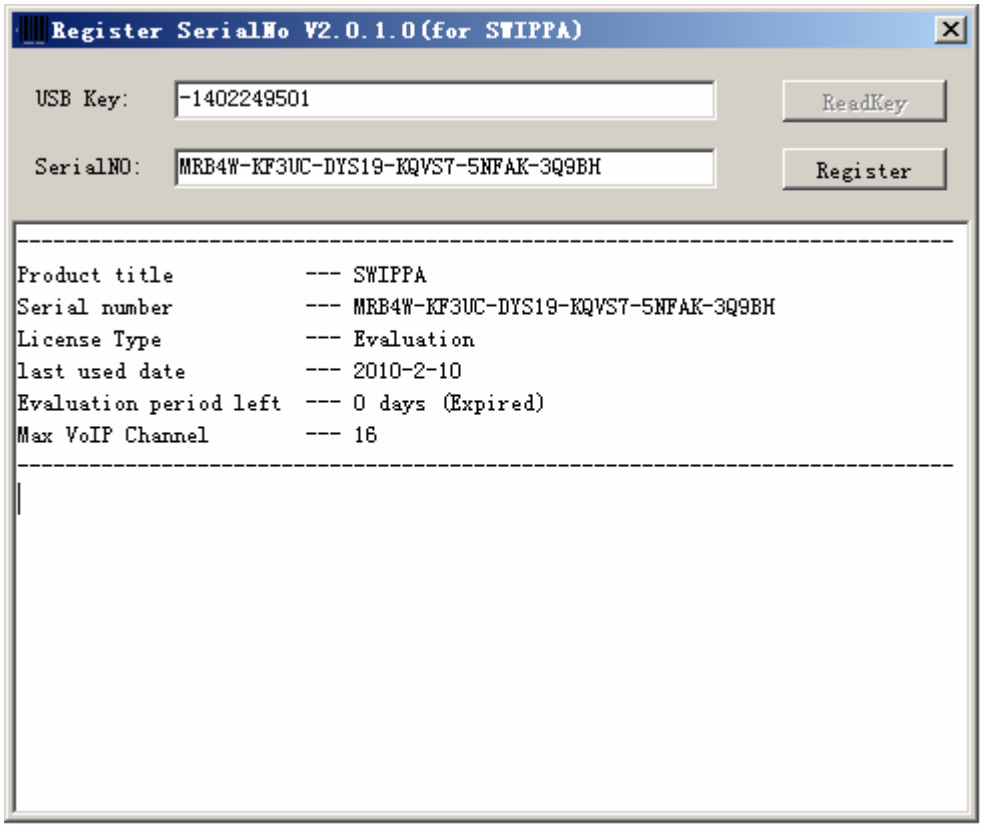


Click button <New Register>.



Click button <ReadKey> to activate USB Key, then enter serial number and click button

<Register>, if authorization code is entered correctly, then the Window below is shown.



You can get authorization code from Synway or Synway authorized partners.

4 Configuration

After installation, following configuration items shall be set for SWIPPA Server.

Number	Configuration item	Section	Description
1	PluginName	System	Plugin name, each plugin is provided for each VOIP PBX
2	RecordNIC	NIC	Name of recording network card
3	StoreSvrIP	StoreSvrID=1	IP address of storage server, make sure to use authentic IP address. It is recommended not to use local IP address 127.0.0.1. If recording application and IPPA server are installed on multiple servers, then recorded voice stream may be wrong

Original setting is recommended for other configuration items.

Configuration information is stored in the file SWIPPAs.ini under installation directory of SWIPPA runtime package.

4.1 [SWIPPAs.log]

Number	Configuration item	Valid value range	Description
1	LogRollerCount	Default value is 5 Valid range: is [0,10000]	Amounts of log files
2	LogSize	Default value is 512 Valid range is [0,10000]	Size (KB) of log file

[SWIPPAs.log]

;Amount of log file

LogRollerCount=5

;Size of log file

LogSize=512

4.2 [System]

Number	Configuration item	Valid value range	Description
1	CheckRtpTimeoutIntval	Default value is 15 Valid range is [0,10000]	Advanced setting, the item is used to set RTP overtime interval in seconds
2	IPPASvrPort	Default value is 10700 Valid range is [1,65535]	IP port of SWIPPA Service
3	PluginName	-	Name of loaded plugin
4	StoreSvrCount	Default value is 1	Amounts of storage server

[System]

; advanced setting

CheckRtpTimeoutIntval=15

; advanced setting

vlMgrLogLevel=4

; dvanced setting

IPPASvrPort =10500

; advanced setting

GCISrvThread=16

; advanced setting

GCISrvCache=16

; advanced setting

AppDealThreads=16

; advanced setting

QueueSize=8192

; advanced setting

DeviceID=71

0: VOIP protocol control message; 1: CTI signaling control; 2: mixed control

ControlMode=0

; Name of loaded plugin

PluginName=ecvlTadiran.dll

; Amounts of storage servers.

StoreSvrCount=1

4.3 [StoreSvrID=x]

Number	Configuration item	Valid range	Description
1	Enable	Default value is 1 Valid range is [0,1]	Whether to enable the service ● 1: enable ● 0: disable
2	StoreSvrIP	IP address	IP address of storage server
3	StoreSvrPort	Default value is 10600	-

[StoreSvrID=1]

; Whether to enable the service

1: enable

0: disable

Enable=1

; IP address of storage server

StoreSvrIP=127.0.0.1

; IP PORT of storage server

StoreSvrPort=10600

; Resource amount

ResoureCounts=300

; Following items are for storage server 2

[StoreSvrID=2]

; Whether to enable the service

1: enable

0: disable

Enable=1

; IP address of storage server

Host=201.123.133.2

; IP PORT of storage server

Port=10600

; Resource amount

ResoureCounts=300

4.4 [Avaya]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65, valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log
3	OutDigit	0~9,*,#	Prefix for outbound calls, if the prefix number is dialed, then event EventBeginRecord will be raised
4	CallCenterNum	Invalid called party number filter list. Semi-colon is used to separate multiple numbers.	Invalid called party number filter list. If called party number is included in the list, then event EventUpdateStatus will not be raised
5	telephonists	Operator channel number list. Semi-colon is to separate multiple channel numbers.	If it is recognized as recording for an operator, special procedure is required Semi-colon is to separate multiple channel numbers

[Avaya]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation

RtpTimeout=65 *;Min is 10, Default is 65 Seconds*

;To set enable/disable channel debugging log

Debug=TRUE

;Prefix for outbound calls, if the prefix number is dialed, then event EventBeginRecord will be raised.

OutDigit=9

;Invalid called party number filter list. If called party number is included in the list, then event EventUpdateStatus will not be raised. Semi-colon is used to separate multiple numbers.

CallCenterNum=XXXXXX;

;Operator channel number list, it is used to determine whether additional process is required after recording of an operator is over. Semi-colon is used to separate multiple channel numbers.

telephonists=0;

;advanced configuration

NeedAgentID=FALSE

;advanced configuration

ECCSrvIP=127.0.0.1

;advanced configuration

ECCSrvTLinksPortMin=1039

;advanced configuration

ECCSrvTLinksPortMax=1039

4.5 [Alcatel]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65 Valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log

[Alcatel]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation

RtpTimeout=65

;To set enable/disable channel debugging log

Debug=TRUE

; advanced configuration

FiltNum=2010

4.6 [Cisco]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65 Valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log
3	GWAddr	Address of Cisco Call Manager	Address of Cisco Call Manager
4	MonitorMode	Default value is 1 Valid range is [0,1]	0: extension number based tapping mode is used. 1: IP address based tapping mode is used.

[Cisco]

; RTP timeout setting. The event EventEndRecord will be raised if there is no RTP package during the period of time set by this item in a conversation.

RtpTimeout=65

; To set enable/disable channel debugging log

Debug=TRUE

; Address of Cisco Call Manager

GWAddr=192.168.1.2;

;To set tapping mode.

0: extension number based tapping mode is used.

1: IP address based tapping mode is used.

MonitorMode=1;

;If extension number based tapping mode is used, then display name of IP phone must be set to the extension number.

4.7 [H4k]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65 Valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log

Number	Configuration item	Valid range	Description
3	MonitorMode	Default value is 1 Valid range is[0,1]	• 0: extension number based tapping mode is used. • 1: IP address based tapping mode is used.

[H4k]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation

RtpTimeout=65;

;To set enable/disable channel debugging log

Debug=TRUE;

;To set tapping mode.

0: extension number based tapping mode is used.

1: IP address based tapping mode is used.

MonitorMode=1

; advanced configuration

IPNumHeader=6

; advanced configuration

IPNumLength=3

4.8 [H8K]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65 Valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch • TRUE—enable the log • FALSE—disable the log
3	MirorPoint	Default value is 0 Valid range is [0,1]	• 1: mirror of SBC (Session Border Controllers) • 0: mirror of H8KSIP server
4	GWAddr	IP address	• If MirorPoint is set to 1, then GWAddr should be set to IP address of SBC • If MirorPoint is set to 0, then GWAddr should be set to IP address of H8K SIP server

Number	Configuration item	Valid range	Description
5	H8KAddr	IP address	• If MirorPoint is set to 1, then the item should be set to address of H8K SIP server • If MiroPoint is set to 0, then the item is reserved
6	MonitorMode	Default value is 1 Valid range is [0,1]	• 0: extension number based tapping mode is used • 1: IP address based tapping mode is used
7	SipPort	Default value is 5060 To set IP port	To set IP port for SIP calls

[H8K]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation

RtpTimeout=65

; To set enable/disable channel debugging log

Debug=TRUE

; 1: mirror of SBC (Session Border Controllers); 0: mirror of H8KSIP server

MirorPoint =0

If MirorPoint is set to 1, then GWAddr should be set to IP address of SBC

If MirorPoint is set to 0, then GWAddr should be set to IP address of H8K SIP server.

GWAddr=192.168.1.2

; If MirorPoint is set to 1, then the item should be set to address of H8K SIP server.

; If MiroPoint is set to 0, then the item is reserved.

H8KAddr=192.168.1.3

; To set IP port for SIP calls

SipPort=5060

; 0: extension number based tapping mode is used.

; 1: IP address based tapping mode is used.

MonitorMode=0

; advanced configuration

CTSAddr=127.0.0.1

; advanced configuration

CTSPort=3001

4.9 [Huawei]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65 Valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log
3	GWAddr	Address of Cisco Call Manager	Address of SIP server
4	MonitorMode	Default value is 1 Valid range is [0,1]	● 0: extension number based tapping mode is used ● 1: IP address based tapping mode is used
5	SipPort	IP port, with default value of 5060 IP	To set IP port for SIP calls

[Huawei]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation

RtpTimeout=65

; To set enable/disable channel debugging log

Debug=TRUE

; Address of HW SIP server

GWAddr=192.168.1.2

; To set IP port for SIP calls

SipPort=5060

; advanced configuration

CTSAddr=127.0.0.1

; advanced configuration

CTSPort=3001

4.10 [Mitel]

Number	Configuration item	Valid range	Description

1	RtpTimeout	Default value is 65 Valid range [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log
3	GWAddr	Address Cisco Call Manager	Address of media gateway

[Mitel]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation

RtpTimeout=65

; To set enable/disable channel debugging log

Debug=TRUE

; AddressSWIPPA Quick Star of Mitel server

GWAddr=192.168.1.2

4.11 [Tadiran]

Number	Configuration item	Valid range	Description
1	RtpTimeout	Default value is 65 Valid range is [0,10000]	Set RTP timeout in seconds. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation
2	Debug	TRUE	To set channel debugging log switch ● TRUE—enable the log ● FALSE—disable the log
3	GWAddr	Address of Cisco Call Manager	Address of media gateway

[Tadiran]

; Set RTP timeout. Event EventEndRecord is raised when no RTP is received during the period of time set by this item in a conversation.

RtpTimeout=65

; To set enable/disable channel debugging log

Debug=TRUE

; Address of Tadian server

GWAddr=192.168.1.2

4.12 [NIC]

Number	Configuration item	Valid value	Description
1	RecordNIC	Name of recording network card	Name of recording network card
2	NIC1	Name of recording network card	The item is updated automatically every time when SWIPPA Server is started
3	NIC2	Name of recording network card	The item is updated automatically every time when SWIPPA Server is started

[NIC]

; Name of recording network card , and recording network card connects with mirror port of the Switch

RecordNIC= Realtek RTL8169/8110 Family Gigabit Ethernet NIC

NIC1= Realtek RTL8169/8110 Family Gigabit Ethernet NIC

NIC2=Realtek RTL8169/8110 Family Gigabit Ethernet NIC

5 Debugging

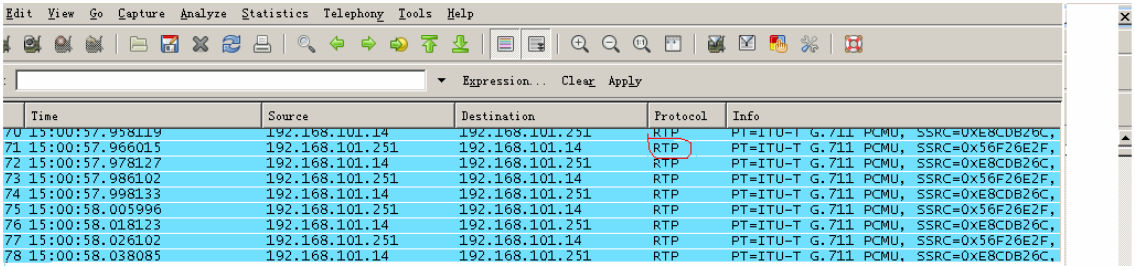
Wireshark® is used for VoIP recording debugging purpose.

Wireshark® is the world's most popular network protocol analyzer. It has a broad range of powerful features and runs on most operating systems including Windows, OS X, Linux, and UNIX. Network professionals, security experts, developers, and educators around the world use it regularly. It is freely available as open source, and is released under the GNU General Public License version 2.

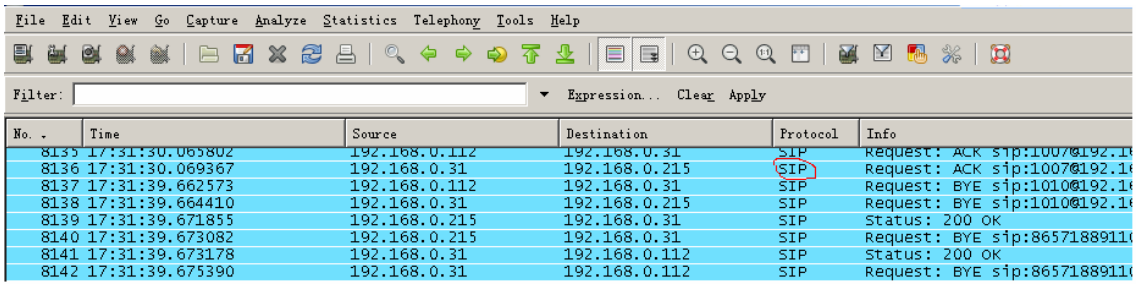
Wireshark® used to be known as Ethereal®. If you're still using Ethereal, it is strongly recommended that you upgrade to Wireshark®.

Please use the current version or you can visit http://www.wireshark.org to download the latest one.

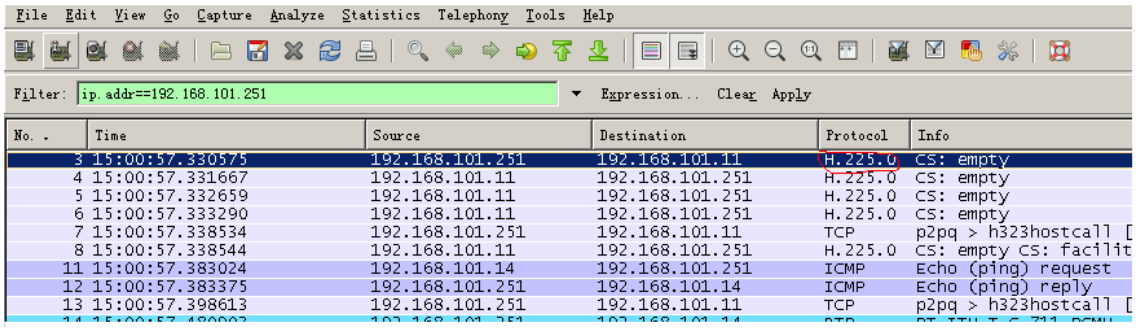
5.1 RTP Package Identification



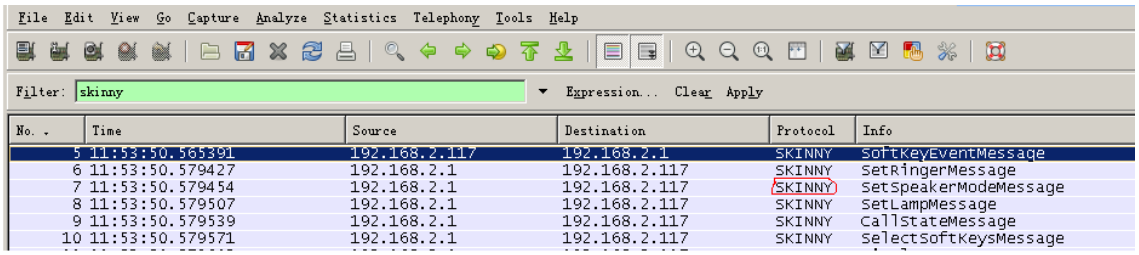
5.2 SIP Package Identification



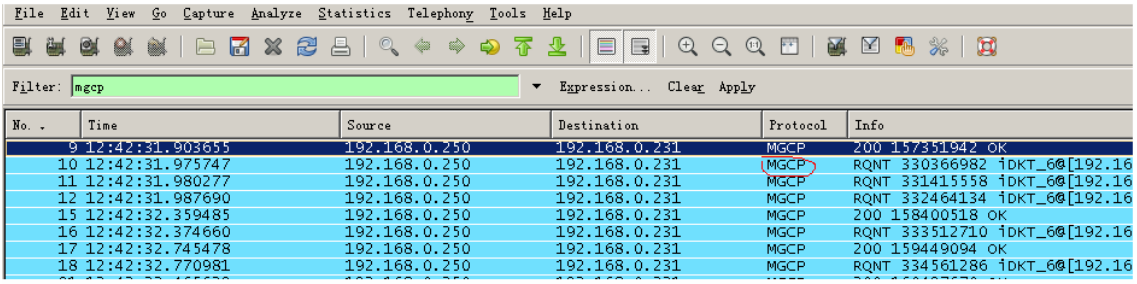
5.3 H323 Package Identification



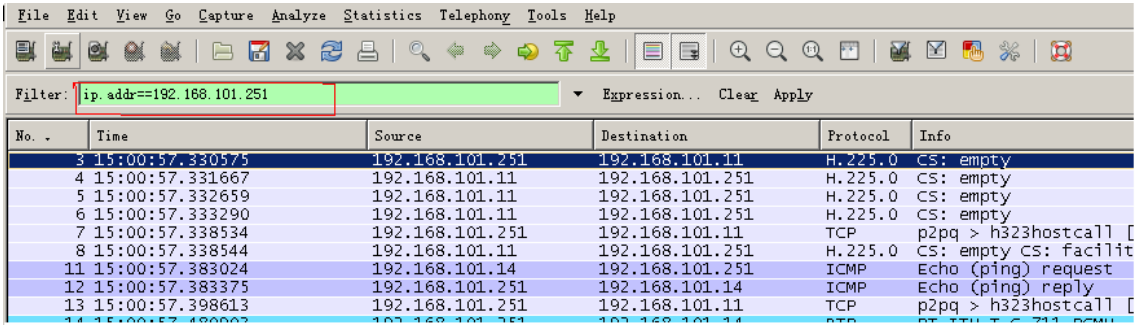
5.4 SCCP Package Identification



5.5 MGCP Package Identification



5.6 IP Address Filter



6 FAQ

1. IP package of tapped IP phone is not mirrored to recording server correctly

(1) Either signaling package or RTP package is not mirrored.

(2) Neither signaling package nor RTP is mirrored.

Solution:

Ask the network engineer to check setting of mirror port of the Switch .

2. Signaling encryption

IP package information can not be checked via wireshark®.

Solution:

Ask IP switch engineer to disable encryption function of IP Switch.