

Современное отечественное решение ВКС Vinteo



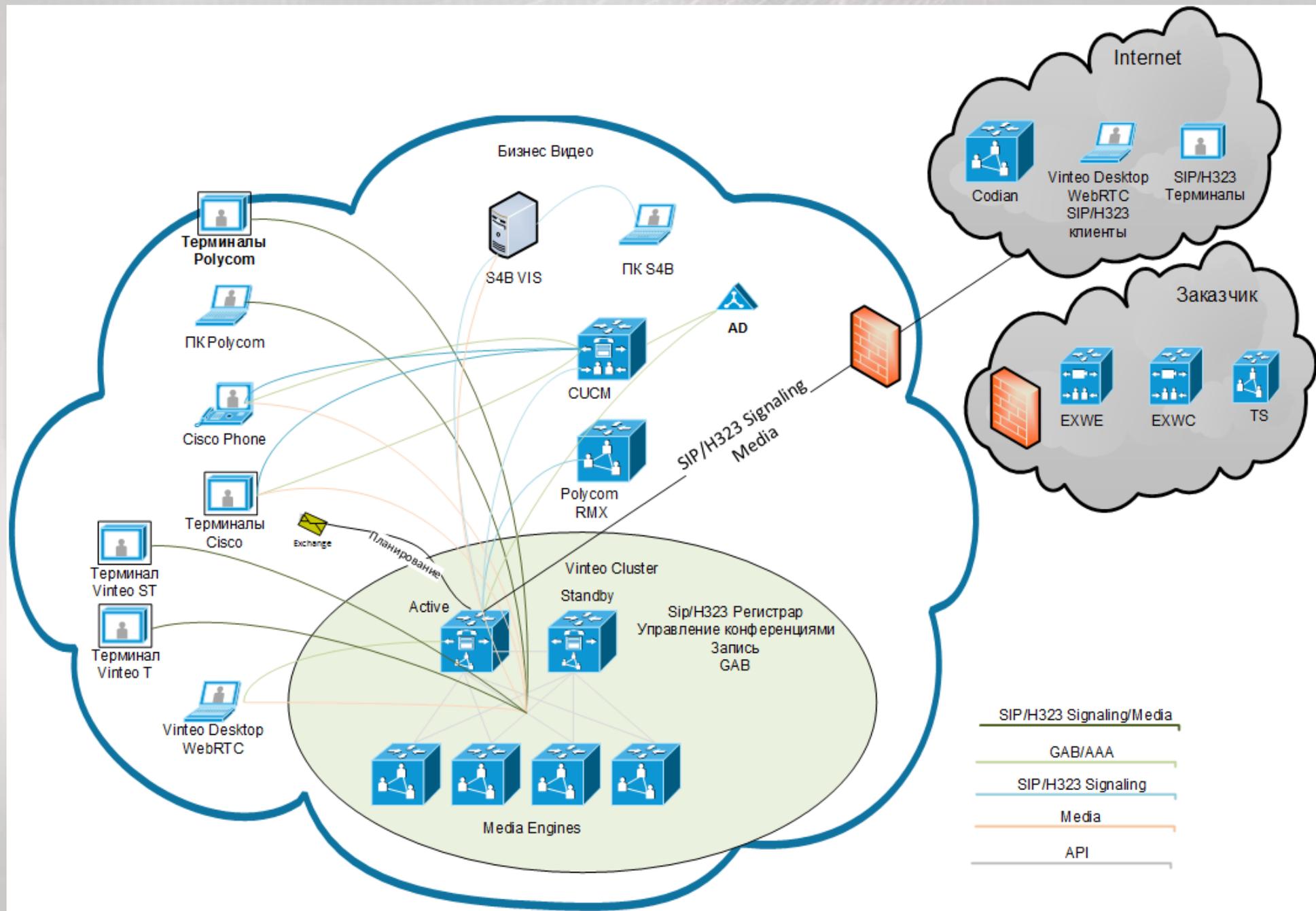
Интеграция Vinteo с
инфраструктурой зарубежных
производителей: Cisco, Polycom,
Microsoft

Видеорешения без компромиссов
Теперь ВКС подстраивается под вас

Интеграция

- Общая схема интеграции Vinteo с системами зарубежных производителей
- Cisco Codian & TMS
- Cisco ExpE & ExwC
- Cisco CUCM & CMS
- Polycom (Plantronics)
- S4B
- Сторонние продукты
- LDAP
- Запись и трансляция

- Общая схема интеграции Vinteo с системами зарубежных производителей



• Cisco CUCM SIP Profile

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Profile. The browser address bar shows the URL: `https://192.168.10.5/ccmadmin/sipProfileEdit.do?key=26409b1f-d0d5-737b-fa26-f2cb2d6deee9`. The page title is "SIP Profile Configuration".

Navigation: System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Related Links: Back To Find/List

Actions: Save, Delete, Copy, Reset, Apply Config, Add New

Status:
• Status: Ready
• All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information:

- Name*: Standard SIP Profile for MCU Vinteo
- Description: Standard SIP Profile for MCU Vinteo
- Default MTP Telephony Event Payload Type*: 101
- Early Offer for G.Clear Calls*: Disabled
- User-Agent and Server header information*: Send Unified CM Version Information as User-Agen
- Version in User Agent and Server Header*: None
- Dial String Interpretation*: Phone number consists of characters 0-9, *, #, anc
- Confidential Access Level Headers*: Disabled

Redirect by Application
 Disable Early Media on 180
 Outgoing T.38 INVITE include audio mline
 Use Fully Qualified Domain Name in SIP Requests
 Assured Services SIP conformance

SDP Information:

- SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*: TIAS and AS
- SDP Transparency Profile: Pass all unknown SDP attributes
- Accept Audio Codec Preferences in Received Offer*: Default
- Require SDP Inactive Exchange for Mid-Call Media Change
- Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone:

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10

Taskbar: Пуск, Internet Explorer, Google Chrome, Skype, 18:37 22.12.2016

• Cisco CUCM

SIP Profile Configuration

Navigation: Cisco Unified CM Administration

Parameters used in Phone:

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000

• Cisco CUCM

The screenshot shows the Cisco Unified CM Administration interface for SIP Profile Configuration. The browser address bar shows the URL: `http://192.168.10.5/ccmadmin/sipProfileEdit.do?key=26409b1f-d0d5-737b-fa26-f2cb2d6deee9`. The page title is "SIP Profile Configuration".

Navigation: Cisco Unified CM Administration | alex | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Related Links: Back To Find/List

Save | Delete | Copy | Reset | Apply Config | Add New

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Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000

Taskbar: SIP Trunk Security...html | Trunk Configuration.html | SIP Profile Configu...html | Показать все

System tray: EN | 18:40 | 22.12.2016

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Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000

The Windows taskbar at the bottom shows the Start button, taskbar icons for Internet Explorer, Google Chrome, and other applications, and a system tray with the date and time: 18:40, 22.12.2016.

• Cisco CUCM SIP Trunk

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The browser address bar shows the URL: `http://192.168.10.5/ccmadmin/trunkEdit.do?key=461e4eee-700e-a277-20aa-4e7192fe95ff`. The page title is "Trunk Configuration".

Navigation: System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration: Save | Delete | Reset | Add New

Status: Status: Ready

SIP Trunk Status: Service Status: Unknown, Duration: Unknown

Device Information:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type:	None(Default)
Device Name*:	Trunk_to_MCU_Vinteo
Description:	Trunk_to_MCU_Vinteo
Device Pool*:	Astana_DP
Common Device Configuration:	< None >
Call Classification*:	Use System Default
Media Resource Group List:	< None >
Location*:	Astana_Location
AAR Group:	< None >
Tunneled Protocol*:	None
QSIG Variant*:	No Changes
ASN.1 ROSE OID Encoding*:	No Changes
Packet Capture Mode*:	None
Packet Capture Duration:	0

Media Termination Point Required

Retry Video Call as Audio

Path Replacement Support

Transmit UTF-8 for Calling Party Name

Transmit UTF-8 Names in QSIG APDU

Unattended Port

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*
When using both sRTP and TLS

Route Class Signaling Enabled*
Default

Use Trusted Relay Point*
Default

PSTN Access

Run On All Active Unified CM Nodes

EN 18:42 22.12.2016

• Cisco CUCM

Trunk Configuration | Управление конференциями

https://192.168.10.5/ccmadmin/trunkEdit.do?key=461e4eee-700e-a277-20aa-4e7192fe95ff

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | alex | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration

Save | Delete | Reset | Add New

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*

Use Trusted Relay Point*

PSTN Access

Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

MLPP and Confidential Access Level Information

MLPP Domain

Confidential Access Mode

Confidential Access Level

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type*

SIP Privacy*

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="< None >"/>	<input checked="" type="checkbox"/>

Incoming Called Party Settings

SIP Trunk Security... | Trunk Configuration.html | SIP Profile Configu... | Показать все

Пуск | 18:43 22.12.2016

• Cisco CUCM

Trunk Configuration | Управление конференцией

https://192.168.10.5/ccmadmin/trunkEdit.do?key=461e4eee-700e-a277-20aa-4e7192fe95ff

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Trunk Configuration | Related Links: Back To Find/List

Save | Delete | Reset | Add New

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS: < None >

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS: < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*: Originator

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Calling and Connected Party Info Format*: Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS: < None >

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN:

Caller Name:

Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Trunk Security... | Trunk Configuration.html | SIP Profile Configu... | Показать все

Пуск | 18:43 22.12.2016

• Cisco CUCM

Save Delete Reset Add New

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration	
1*	<input type="text" value="192.168.10.26"/>	<input type="text"/>	<input type="text" value="5060"/>	N/A	N/A	N/A	<input type="button" value="+"/> <input type="button" value="-"/>

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

• Cisco CUCM

Save | Delete | Reset | Add New

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1* 192.168.10.26		5060	N/A	N/A	N/A

MTP Preferred Originating Codec*

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Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

Parameter Name	Parameter Value
1	

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None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

Вопросы?

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