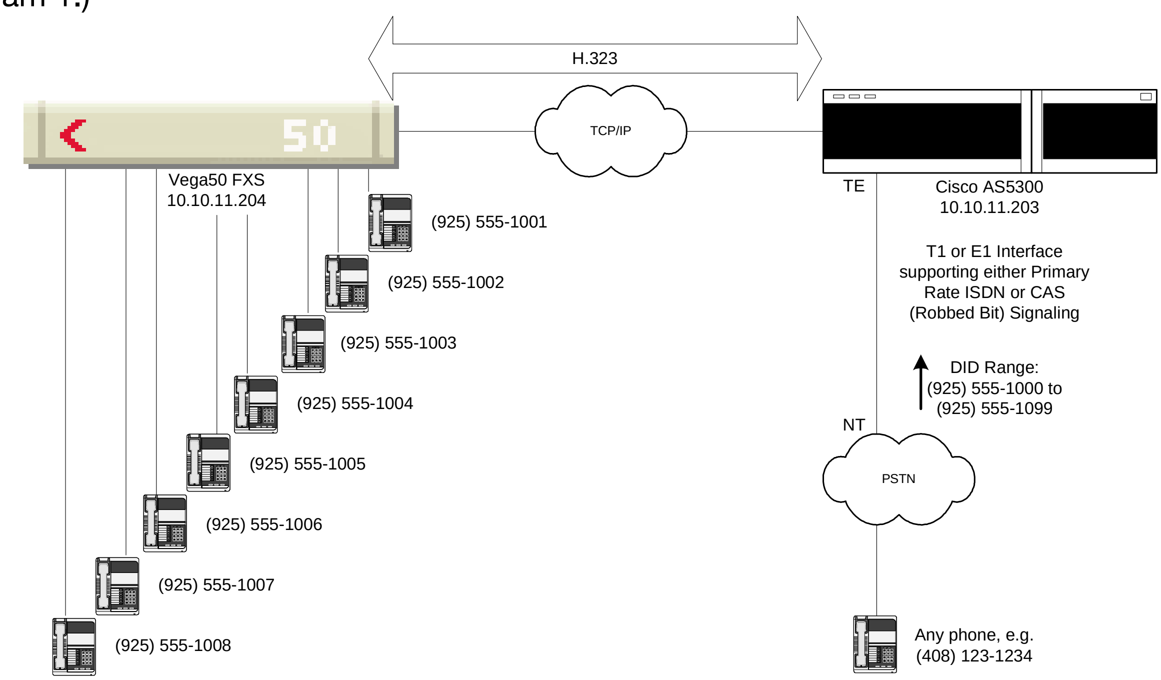
Interoperation Guide

R4 to R7 Vega to Cisco AS5300 and AS2600

Purpose and Overview:

The purpose of this document is to describe the steps necessary to establish and configure a distributed voice communication solution utilizing the VegaStream line of products and the Cisco AS5300 voice gateway.

(Diagram 1.)



This document covers the configuration required for Vega and Cisco systems where the Vega is using Release 4 or higher H.323 code.

The main section of the document describes the specific parameters that are needed to make the Vega and the Cisco gateways interwork. It assumes that the gateways are already configured correctly on their LAN and Telephony interfaces.

A number of annexes are also included as follows:

Configuring Vega and Cisco AS5300 to interwork

The diagram on Page 1 (Diagram 1.) illustrates a basic distributed communication solution. The Cisco AS5300 is located at a central location and is interfaced to the PSTN using any of the methods supported by the AS5300; T1, E1, Primary Rate ISDN (PRI) signaling, CAS (Robbed Bit) signaling.

The VegaStream products (the unit used in this example is a Vega50 FXS) are placed at remote locations and are connected to the Cisco AS5300 across an IP network; either a managed IP network or the public Internet.[1] The call control engines that are integrated in to the Vega50 FXS and the Cisco AS5300 are then both configured to create voice links between the two points.

Each analog handset[2] connected to the Vega50 FXS is associated with either a directory number (E.164 number) or a short extension. In this example, a full E.164 number is used and this number corresponds to a direct inward dial (DID) number available from the PSTN trunk lines interfaced to the AS5300. This example, as illustrated in the diagram above, shows a single configuration scenario; however, there are a great number of options available from this basic solution design. For example, the IVR capabilities available from the AS5300 could be utilized instead of DID, or DID could be utilized in several different ways

[1] The overall system performance for this solution will be dependant on the overall viability of the IP network. It is recommended that as robust a network as possible be utilized and that sufficient bandwidth over the whole length of the network is available.

[2] It would be equally as viable a solution to use the Vega50 FXS to provide analog trunk lines to a PBX. This solution would be transparent to the PBX, as the trunk lines would appear no different to the PBX than analog trunk lines originating from the PSTN, and all of the features and capabilities of the PBX could be utilized, such as auto-attendant, voice mail, IVR and automatic call distribution (ACD) systems. Additional features not covered in this document are available from the VegaStream product line to enhance this configuration, such as call representation.

with or without the IVR features. Offering an explanation of every possible configuration is well beyond the scope of this document, but it is good to be aware that the solution described is very flexible and scalable, and that many additional features of both the VegaStream product line and the AS5300 could be utilized for variations of this basic solution design. For the duration of this document, configuration examples will be limited to the scenario illustrated above and it will be assumed a Vega50 FXS unit is being used.

In the example above, when a call to one of the DID numbers associate with the Cisco AS5300 trunk lines originates from the PSTN, the AS5300 will route the incoming call across the IP network to the Vega50 FXS and the Vega50 FXS will route the call to the appropriate handset. When the handset connected to the Vega50 FXS is taken off-hook, the call is connected across the entire route. In the same way, when a handset connected to the Vega50 FXS is taken off-hook and a PSTN telephone number dialed, the Vega50 FXS passes the information to the AS5300 which in turn passes the call across one of its trunk lines to the PSTN. When the dialed party on the PSTN answers the call, the call is connected across the entire route.

Following the example above, if the telephone handset with the number (408) 123-1234 (a telephone directly connected to the PSTN) is taken off-hook and the number (925) 555-1003 is dialed, the handset connected to the Vega50 FXS labeled (925) 5551003 will ring and the call will be connected when it is taken off-hook. The same is true if the call originates from the handset connected to the Vega50 FXS. In short, the end user experience is identical to that of a traditional PSTN telephone call.

Additionally, both the VegaStream product line and the Cisco AS5300 have standards based facilities for managing billing (CDR records) and network monitoring/management (SNMP MIB1 & MIB2).

Creating a Network Plan:

Prior to configuring your Vega50 FXS and/or Cisco AS5300, it is a good practice to first plan and diagram your network and numbering plan. Although it is safe to assume that your network will be considerably more complex than the diagram provided as an example with this document, a similar diagram will prove to be a good place to start.

As we begin configuring the Vega50 FXS and the Cisco AS5300 to manage call flow across the network, it will be necessary to have a pre-determined and compiled list of the IP addresses of the various network devices as well as the telephone numbering scheme (E.164, short extensions, . . . etc.). It will be necessary to have this information, as both the Vega50 FXS and the AS5300 will require that this information be entered in to the configuration for call control. Not having this well planned before hand and trying to make things up as you go can prove to make things very complicated and inefficient, and often results in errors in your configuration.

It is also a good idea when planning or expanding your communications network to estimate the amount of network bandwidth that will be necessary to accommodate the level of anticipated traffic. It is possible that you will discover that you network may need expanding to accommodate traffic increases as your network expands.

Additionally, when making any changes or additions to your communications network, it is good practice to document and save all relevant information and documentation for the system before and after the change. VegaStream products and the AS5300 allow you to save the system configurations to files that can be backed up, archived and used to restore the system if necessary. Taking these precautions makes the process of restoring your network, troubleshooting issues and making future changes much easier.

Cisco AS5300 Configuration:

The purpose of this section is not to provide a detailed step-by-step description describing how to completely configure every aspect of your Cisco AS5300, but a general explanation of what is required for these products to work together. Although the process of configuring the call flow controls is described in detail, such things as the configuration of the AS5300 interface to the PSTN is not covered in detail. The number of possible variables and the complexity of such a task is well beyond the scope of this document. For example, the AS5300 supports both T1 and E1 interfaces and PRI and CAS signaling. To provide instruction for this configuration and all the possible variables, and to keep these instructions up to date is a task more suited directly for Cisco. For any configuration documentation and instruction not covered in this document, please refer to the Cisco web site (www.Cisco.com), or contact Cisco technical support.

This document makes the following assumptions regarding the configuration of the Cisco AS5300:

- **Cisco AS5300 IOS Version:** This document assumes that the version of the AS5300 IOS software installed is (show version) at least:
  - IOS 5300 Software (C5300-IS-M), Version 12.2(3), RELEASE SOFTWARE.
  If T.38 operation is required then the version of IOS must be at least:
  - IOS 5300 Software (C5300-IS-M), Version 12.2(11).
  If this is not the version of the IOS software currently installed on your AS5300, please refer to the following for upgrade information:

  http://www.cisco.com/warp/public/130/sw_upgrade_proc_ram.shtml

  This document assumes that the version of the VCware (the software necessary to operate the AS5300 Voice Cards) installed is (show vfc x version vcware [x=0|1|2]): VCware Version: 7.35, ROM Monitor Version: 1.2, DSPware Version: 3.4.46L Technology: C542.
  To upgrade your VCWare, please refer to the following for upgrade information[3]:

[3] The version of the VCWare software you need will depends on what version of Voice Card(s) you have installed in your Cisco AS5300. It is quite important that that you have the software that corresponds to your hardware in order for your

http://www.cisco.com/univercd/cc/td/doc/product/access/acs_serv/5300/iosrn/vcwrn/rnvcw8xx.htm

- **IP Configuration:** This document assumes that the configuration of the AS5300 as a node on the IP network(s) it is connected to is complete and correct.
- **IP Routing:** This document assumes that if the AS5300 is being used for the routing of IP traffic, in addition to voice traffic, that the appropriate routing tables have been configured and are correct.

  Note: Even if the AS5300 is not being used to route IP traffic, IP Routing must be enabled and a default route entered in order for voice traffic to be correctly routed. Simply assigning a default gateway will not work.
- **PSTN Interfaces:** This document assumes that all interfaces to the PSTN (T1, E1, PRI or CAS signaling) have been configured, are correct and are functioning as necessary to support the desired call controls.

  Note: The **voice-port bearer-cap** should be set to **3100Hz** for voice calls.
- **VoIP Protocol:** This document assumes that the ITU-T H Series Recommendation H.323 version 2 will be the VoIP protocol implemented for this solution. At the time of publication, this is the only VoIP protocol supported by the Cisco AS5300 IOS software.

Cisco AS5300 Voice Service Configuration:

Once the basic configuration, as outlined above, is complete, the process of configuring the AS5300 for operation with the Vega50 FXS is remarkably straightforward. In short, it requires configuring two dial-peers and two voice classes. The entire process should take less than 20 minutes. Beginning from the global configuration mode (See Annex 6), complete the following steps:

Configuring the dial-peer for the PSTN

Step 1	Command	Description
1.1	Router (config) # **dial-peer voice 1 pots**	Enter dial-peer configuration mode in order to create a dial-peer to handle calls to/from the PSTN.
1.2	Router (config-dial-peer) # **destination-pattern .**	Sets the **destination-pattern** of calls that this dial peer will handle (to send calls to the PSTN interface). In this case, it sets this dial peer to accept any number (indicated by the "**.**").
1.3	Router (config-dial-peer) # **direct-inward-dial**	Enables the direct-inward-dial (DID) capability. This parameter

system to operate correctly. If you have any questions regarding what hardware you have installed and/or what version of the software you will need, please contact Cisco technical support or refer to the Cisco web site: www.Cisco.com.

		is enabled when the interface to the PSTN supports DID and this feature is to be used for incoming call routing.
1.4	Router (config-dial-peer) # **incoming called-number .**	Defines the **incoming called-number**(s) to accept from the PSTN interface. In this case **incoming called-number** can be any number (indicated by the ".")
1.5	Router (config-dial-peer) # **port 0:D**	Specify the Voice-port parameters to use (see below for configuration of voice-port).
1.6	Router (config-dial-peer) # **no register e164**	Disables the registration of E.164 numbers
1.7	Router (config-dial-peer) # **exit**	Exits the dial-peer configuration mode

Configuring a voice-class for codec selection

Step 2	Command	Description
2.1	Router (config) # **voice class codec 1**	Creates the voice class and enters the class configuration mode
2.2	Router (config-class) # **codec preference 1 *g729r8***	Sets the G.729 Annex A audio codec and preference 1
2.3	Router (config-class) # **codec preference 2 *g729br8***	Sets the G.729Annex B audio codec and preference 2
2.4	Router (config-class) # **codec preference 3 *g711ulaw***	Sets the G.711uLaw audio codec and preference 3
2.5	Router (config-class) # **codec preference 4 *g711alaw***	Sets the G.711 aLaw audio codec and preference 4
2.6	Router (config-class) # **codec preference 5 *g723r63***	Sets the G.723.1 audio codec and preference 5
2.7	Router (config-class) # **exit**	Exits the class configuration mode

Note: VegaStream products support all the audio compression codecs listed above; G.729, G.729 Annex A, G.711uLaw, G.711aLaw and G.723.1. The codec preferences can be in any order, not necessarily the same order as above, and undesirable codecs can be excluded from the list of available codecs both on the Cisco AS5300 and on VegaStream products

Configuring a voice-class for H.323 parameters

Step 3	Command	Description
3.1	Router (config) # **voice class h323 1**	Creates the voice class and enters the class configuration mode
3.2	Router (config-class) # **call start fast**	Sets H.323 calls to fast start.
3.3	Router (config-class) # **exit**	Exits the class configuration mode

Configuring the dial-peer for VoIP

Step 4	**Command**	**Description**
4.1	Router (config) # **dial-peer voice 2 voip**	Enter dial-peer configuration mode in order to create a dial-peer to handle calls to/from the VoIP interface.
4.2	Router (config-dial-peer) # **incoming called-number .**	Defines the **incoming called-number**(s) to accept from the VoIP interface. In this case **incoming called-number** can be any number (indicated by the ".")
4.3	Router (config-dial-peer) # **destination-pattern *55510..***	Sets the **destination-pattern** of calls that this port can handle (to send to the VoIP interface). In this case, it sets this dial peer to accept any number beginning "**55510**" and ending with any two digits (indicated by the "**..**" variable). [Note, this also causes this dial peer to accept inbound calls with an ANI 55510..]
4.4	Router (config-dial-peer) # **session target ipv4: *x.x.x.x***	The actual IP address would be the IP address of the Vega50 FXS, e.g. ipv4:10.10.11.204
4.5	Router (config-dial-peer) # **voice-class codec 1**	Associates the **voice-class codec 1** with this dial-peer. The properties of this voice-class will be adopted by this dial-peer.
4.6	Router (config-dial-peer) # **voice-class h323 1**	Associates the **voice-class h323 1** with this dial-peer. The properties of this voice-class will be adopted by this dial-peer.
4.7	Router (config-dial-peer) # **dtmf-relay h245-signal**	Enables DTMF relay using H.245 signaling
4.8	Router (config-dial-peer) # **no vad**	Turns off Voice Activity Detection (VAD)
4.9	Router (config-dial-peer) # **exit**	Exits the dial-peer configuration mode

Configuring the Cisco trunks to force bearer capability value on ISDN trunk

Step 5	**Command**	**Description**
5.1	Router (config) # **voice-port 0:D**	Configure port settings
5.2	Router (config-voiceport) # **bearer-cap 3100hz**	Set bearer capability for trunk 1 to 3.1khz audio
5.3	Router (config-voiceport) # **exit**	

Following the completion of the necessary configuration for voice service, it is a good idea to save the changes and back-up the configuration.

Step 6	Command	Description
6.1	Router (config) # **exit**	Exit out of configuration mode
6.2	Router # **write memory**	Write configuration to internal hard disc
6.3	Router # **reload**	Reboot system
6.4	Router # **copy system:/rumning-config tftp:*filepath/filename***	Backup configuration to tftp server

Vega Configuration:

As with the configuration instructions for the Cisco AS5300, this document concentrates on covering the VegaStream configuration parameters relevant to the interoperation of these two products to provide the voice service described earlier in this document. Additional documentation and instructions are provided with your VegaStream product. If additional assistance is required, please contact VegaStream Technical Support[4]. This document makes the following assumptions:

- **VegaStream firmware Version for H.323:** This document assumes that the version of the VegaStream firmware is (show version) at least: R4 T028-123 for Vega 50 (R4 T028-12 for Vega 100). If the version of the firmware currently installed on your VegaStream product needs upgrading, you may obtain the latest firmware and upgrade procedure from: www.VegaAssist/firmware
- **Vega Product:** The configuration instruction provide in this document is specifically for the Vega50 FXS H.323 product. While the configuration of all VegaStream products is similar, and various VegaStream products could be used in voice communication solutions similar to that described in this document, the instructions in this document are only accurate for the Vega50 FXS product. For assistance configuring other VegaStream products please contact VegaStream Technical Support.
- **IP Configuration:** This document assumes that prior to configuring the voice features of the Vega50 FXS, that the configuration as a node on the IP network it is connected to is complete and correct. For instructions, please consult the documentation provided with your product or contact VegaStream Technical Support.

Once the basic configuration of the Vega50 FXS is complete, log on to the system as an Administrator for a command line interface (CLI), either from a serial interface terminal or telnet session. Complete the following steps:

Set Caller-Ids for FXS Ports

[4] VegaStream Technical Support can be contacted at support@VegaStream.com. For additional and regional contact information, please refer to our web sites at www.VegaStream.com and www.VegaAssist.com

Step 1	Command	Description
1.1	ADMIN > **set .pots.qslac.codec.1.group.1.dn=*9255551001***	Sets Caller-ID (CLID)
1.2	ADMIN > **set .pots.qslac.codec.2.group.1.dn=*9255551002***	Sets Caller-ID (CLID)
1.3	ADMIN > **set .pots.qslac.codec.3.group.1.dn=*9255551003***	Sets Caller-ID (CLID)
1.4	ADMIN > **set .pots.qslac.codec.4.group.1.dn=*9255551004***	Sets Caller-ID (CLID)
1.5	ADMIN > **set .pots.qslac.codec.5.group.1.dn=*9255551005***	Sets Caller-ID (CLID)
1.6	ADMIN > **set .pots.qslac.codec.6.group.1.dn=*9255551006***	Sets Caller-ID (CLID)
1.7	ADMIN > **set .pots.qslac.codec.7.group.1.dn=*9255551007***	Sets Caller-ID (CLID)
1.8	ADMIN > **set .pots.qslac.codec.8.group.1.dn=*9255551008***	Sets Caller-ID (CLID)

Set H.323 Parameters

Step 2	Command	Description
2.1	ADMIN > **set .h323.use_fast_start=1**	Enable fast start for out going calls
2.2	ADMIN > **set .h323.accept_fast_start=2**	Enable fast start for incoming calls
2.3	ADMIN > **set .h323.h245_after_fast_start=0**	Disable H.245 tunneling after fast start
2.4	ADMIN > **set .h323.use_early_h245=0**	Disable early media for out going calls
2.5	ADMIN > **set .h323.accept_early_h245=0**	Disable early media for incoming calls
2.6	ADMIN > **set .use_h245_tunnel=0**	Disable tunnelling
2.7	ADMIN > **set .accept_h245_tunnel=0**	Disable tunnelling
2.8	ADMIN > **set *advanced.h323*control.signal_dtmf_tx=1**	Use “signal” type messages for Out Of Band DTMF

Set Codec Parameters

Step 3	Command	Description
3.1	ADMIN > **set .h245.cap.1.name=*g729***	Sets G.729 as capability 1
3.2	ADMIN > **set .h245.cap.2.name=*g729AnnexA***	Sets G.729Annex A as capability 2
3.3	ADMIN > **set .h245.cap.3.name=*g711Ulaw64k***	Sets G.711uLaw as capability 3
3.3	ADMIN > **set .h245.cap.4.name=*g711Alaw64k***	Sets G.711ALaw as capability 4
3.3	ADMIN > **set .h245.cap.5.name=*g7231***	Sets G.711uLaw as capability 5
3.4	ADMIN > **set .h245.preferred_index=0**	Offer and accept any of the defined codecs
3.5	ADMIN > **set .h245.voice_capdesc.1.caps=1,2,3,4,5**	Sets preference order for the codecs.
3.6	ADMIN > s**et .dsp.g729.packet_time=20**[5]	Sets packet size for g729 codec
3.7	ADMIN > s**et .dsp.g729AnnexA.packet_time=20**[6]	Sets packet size for g729Annex A codec

[5] 60ms is also acceptable
[6] 60ms is also acceptable

Note: VegaStream products support the following audio compression codecs: G.729, G.729 Annex A, G.711uLaw, G.711aLaw and G.723.1. The example above shows the configuration of G.729, G.729 Annex A and G.711 uLaw. For information on configuring additional codec, please refer to the documentation provided with your VegaStream product of contact VegaStream Technical support. The codec preferences can be in any order, not necessarily the same order as above, and undesirable codecs can be excluded from the list of available codecs both on the Cisco AS5300 and on VegaStream products.

Create Eight New Dial Plans

Step 4	Command	Description
4.1	ADMIN > **profile 1**	
4.2	Planner.profile.1 > **new plan**	
4.3	Planner.profile.1.plan.2 > **new plan**	
4.4	Planner.profile.1.plan.3 > **new plan**	
4.5	Planner.profile.1.plan.4 > **new plan**	Creates 8 new dial plans (9 plans in total)
4.6	Planner.profile.1.plan.5 > **new plan**	
4.7	Planner.profile.1.plan.6 > **new plan**	
4.8	Planner.profile.1.plan.7 > **new plan**	
4.9	Planner.profile.1.plan.8 > **new plan**	
4.10	Planner.profile.1.plan.9 > **cd .**	

Set Parameters of Eight New Dial Plans

Step 5	Command	Description
5.1	ADMIN > **set .planner.profile.1.plan.1.name=Catch1001**	
5.2	ADMIN > **set .planner.profile.1.plan.1.srcc=IF:05,TEL:5551001**	
5.3	ADMIN > **set .planner.profile.1.plan.1.dest=IF:06**	
5.4	ADMIN > **set .planner.profile.1.plan.2.name=Catch1002**	
5.5	ADMIN > **set .planner.profile.1.plan.2.srcc=IF:05,TEL:5551002**	
5.6	ADMIN > **set .planner.profile.1.plan.2.dest=IF:07**	
5.7	ADMIN > **set .planner.profile.1.plan.3.name=Catch1003**	
5.8	ADMIN > **set .planner.profile.1.plan.3.srcc=IF:05,TEL:5551003**	
5.9	ADMIN > **set .planner.profile.1.plan.3.dest=IF:08**	
5.10	ADMIN > **set .planner.profile.1.plan.4.name=Catch1004**	
5.11	ADMIN > **set .planner.profile.1.plan.4.srcc=IF:05,TEL:5551004**	
5.12	ADMIN > **set .planner.profile.1.plan.4.dest=IF:09**	
5.13	ADMIN > **set .planner.profile.1.plan.5.name=Catch1005**	
5.14	ADMIN > **set .planner.profile.1.plan.5.srcc=IF:05,TEL:5551005**	Configures ports on the Vega50 FXS in accordance with the dial plan described earlier in this document.
5.15	ADMIN > **set .planner.profile.1.plan.5.dest=IF:10**	
5.16	ADMIN > **set .planner.profile.1.plan.6.name=Catch1006**	
5.17	ADMIN > **set .planner.profile.1.plan.6.srcc=IF:05,TEL:5551006**	
5.18	ADMIN > **set .planner.profile.1.plan.6.dest=IF:11**	
5.19	ADMIN > **set .planner.profile.1.plan.7.name=Catch1007**	
5.20	ADMIN > **set .planner.profile.1.plan.7.srcc=IF:05,TEL:5551007**	
5.21	ADMIN > **set .planner.profile.1.plan.7.dest=IF:12**	
5.22	ADMIN > **set .planner.profile.1.plan.8.name=Catch1008**	

5.23	ADMIN > **set .planner.profile.1.plan.8.srcc=IF:05,TEL:5551008**	
5.24	ADMIN > **set .planner.profile.1.plan.8.dest=IF:13**	
		Dial plan “DialOut” has an IP address TA:xxxx – in this example xxxx = 10.10.11.203, the IP address of the AS5300
5.25	ADMIN > **set .planner.profile.1.plan.9.name=DialOut**	
5.26	ADMIN > **set .planner.profile.1.plan.9.srcc=IF:.[^5],TEL:<.*>**	
5.27	ADMIN > **set .planner.profile.1.plan.9.dest=IF:05,TEL:<1>,TA:xxxx**	

Save Configuration and Restart System

Step 6	Command	Description
6.1	ADMIN > **save**	Saves the current configuration
6.2	ADMIN > **reboot system**	Reset systems so changes will take effect.

Following the completion of the necessary configuration for voice service, it is a good idea to archive the configuration.

Step 7	Command	Description
7.1	ADMIN > **cp .**	Ensure the Vega is at the root of the path
7.2	ADMIN > **tput *filename.txt***	Archive the configuration to the tftp server

Annex 1 - Fallback to Release 3 Vega settings

For Release 3 Vegas and earlier versions of Cisco code the following configurations may need to be changed:

Faststart may need to be disabled:
On Vega:

	Command	Description
1.1	ADMIN > **set .h323.use_fast_start=0**	Disables fast start for out going calls
1.2	ADMIN > **set .h323.accept_fast_start=0**	Disables fast start for incoming calls
1.3	ADMIN > **set .h323.h245_after_fast_start=0**	Disables H.245 tunneling after fast start
1.4	ADMIN > **set .h323.use_early_h245=0**	Disables early media for out going calls
1.5	ADMIN > **set .h323.accept_early_h245=0**	Disables early media for incoming calls
1.6	ADMIN > **set .use_h245_tunnel=0**	Disable tunnelling
1.7	ADMIN > **set .accept_h245_tunnel=0**	Disable tunnelling

On AS5300:

	Command	Description
2.1	Router (config) # **voice class h323 1**	Creates the voice class and enters the class configuration mode
2.2	Router (config-class) # **call start slow**	Sets H.323 calls to slow start.
2.3	Router (config-class) # **exit**	Exits the class configuration mode

Vega code must be >= R3 T051
Cisco code must be >= Version 12.2(3)

Annex 2 - Troubleshooting

Call clears down with unexpected cause code

Disconnect cause code	Reason
44	Disconnect 44 and 65 can indicate that the bearer capability is not configured correctly on the Cisco gateway. Cisco have documentation on their site at http://www.cisco.com/warp/public/788/signalling/h323-isdn-callfailure.html

47	Disconnect 47 can be observed when using a Vega to make calls to a Cisco endpoint via a Cisco CallManager and tunneling is not enabled. Resolve this on the Vega by enabling h323>tunneling on the web browser or on the CLI interface:
	`set .h323.use_h245_tunnel=1`
	`save`
	`reboot system`

65	Disconnect 44 and 65 can indicate that the bearer capability is not configured correctly on the Cisco gateway. Cisco have documentation on their site at http://www.cisco.com/warp/public/788/signalling/h323-isdn-callfailure.html
88	Check the configuration of Alaw / ulaw on the outgoing ISDN trunk – it may not match the audio coding expected by the PBX / exchange.

One way audio

1. Check the configuration of Alaw / ulaw on the outgoing ISDN trunk – it may not match the audio coding expected by the PBX / exchange.

2. If the VoIP call is using the G.729 codec, and the Vega is configured to use an 80ms packet time (default g729 packet size) one way audio is observed - no audio from the Vega towards the Cisco. Select a better packet time, e.g. 20ms.

Problems with faststart connection

On Vega:

	Command	Description
1.1	ADMIN > **set .h323.use_fast_start=0**	Disables fast start for out going calls
1.2	ADMIN > **set .h323.accept_fast_start=0**	Disables fast start for incoming calls
1.3	ADMIN > **set .h323.h245_after_fast_start=0**	Disables H.245 tunneling after fast start
1.4	ADMIN > **set .h323.use_early_h245=0**	Disables early media for out going calls
1.5	ADMIN > **set .h323.accept_early_h245=0**	Disables early media for

		incoming calls
1.6	ADMIN > **set .use_h245_tunnel=0**	Disable tunnelling
1.7	ADMIN > **set .accept_h245_tunnel=0**	Disable tunnelling

On AS5300:

	Command	Description
2.1	Router (config) # **voice class h323 1**	Creates the voice class and enters the class configuration mode
2.2	Router (config-class) # **call start slow**	Sets H.323 calls to slow start.
2.3	Router (config-class) # **exit**	Exits the class configuration mode

Check Cisco AS5300 setup

AS5300 LAN
Make sure the interface is up ("Ethernet0 is up, line protocol is up")
show interfaces ethernet 0

ping <AS5300 IP address> (using a PC).
ping something from your AS5300.

AS5300 PSTN (T1)
Connect the T1 to the PSTN, you should be getting these or similar messages:

CONTROLLER-5-UPDOWN: Controller T1 0, changed state to up
CSM-5-PRI: add PRI at slot 0, unit 0, channel 23 with index 0
ISDN-6-LAYER2UP: Layer 2 for Interface Se0:23, TEI 0 changed to up
LINK-3-UPDOWN: Interface Serial0:23, changed state to up

Make sure the controller is operational:
show controllers T1 0

AS5300 Dial-peers
Verify each dial-peer:
show dial-peer voice x // x=1,2,….

Notes

Important

- Enable IP routing even if you do not need it, you must enable it.
- VAD (Voice Activity detection) must be disabled for G729.

Check Vega setup

Vega LAN

ping <Vega IP address> (using a PC).
ping something from your Vega

ping the AS5300 from the Vega

To make sure the dial plans are correct, enter **show plan.** It should look like this:

```
Index Grp     Source                                  Destination
/Cost     Int'face Address                        Int'face Address
----- --- -------- ------------------------- -------- ----------------
 1/0  0   05       TEL:5551001                06
 2/0  0   05       TEL:5551002                07
 3/0  0   05       TEL:5551003                08
 4/0  0   05       TEL:5551004                09
 5/0  0   05       TEL:5551005                10
 6/0  0   05       TEL:5551006                11
 7/0  0   05       TEL:5551007                12
 8/0  0   05       TEL:5551008                13
 9/0  0   .[^5]    TEL:<.*>                   05       TEL:<1> TA:10.10.11.203
```

The sources of Plans 1-8 match incoming numbers 555-1001 to 555-1008 and distribute the calls to the appropriate phones.
Plan 9 matches all interface-IDs except those with a "5" at their second digit position (all IDs, except 05,15,25,…). Since there are no interface IDs greater than 13, it matches interface IDs 06..13, which represent the analog FXS-ports. This plan makes Vega50 FXS send the dialed numbers to interface 05, which is the LAN interface.

Test both inbound and outbound calls and make sure you have audio in both directions.

Use `log display on` to check the progress of calls

Annex 3 - Recommended codec configurations by version number

During testing of the interoperation of the Vega with Cisco AS5300s it has been observed that:

If using G.729 compression

- Cisco v12.0 works best with the Vega g729 codec
  - on Cisco configure the g729 r8 codec
  - on Vega configure the G729 parameters, packet_time=20ms[7], VAD=disabled
- Cisco v12.1 works best with the Vega g729AnnexA codec
  - on Cisco configure the g729 br8 codec
- Cisco v12.2 works best with the Vega g729 codec
  - on Cisco configure the g729 r8 codec
  - on Vega configure the G729 parameters, packet_time=20ms[8], VAD=disabled

For G.723.1

- Cisco's g723r63 is equivalent to the Vega g7231 with VADU turned off

The AS5300 does not like to be forced to use just a single codec (so just enabling a single codec or setting preferred index=n (n<>0) is not a valid option) - use the CAPDESC feature to limit the codecs if required (offer at least two codecs).

[7] 60ms is also acceptable
[8] 60ms is also acceptable

Annex 4 - Cisco and Vega cabling

For full details about Cisco cabling the appropriate Cisco documentation should be consulted.

Cisco PRI	Vega 100 PRI
NT (Cisco physical)	**TE** (NT physical)
2 (Rx+)	1 (Tx+)
1 (Rx-)	2 (Tx-)
5 (Tx+)	4 (Rx+)
4 (Tx-)	5 (Rx-)

Cisco PRI	Vega 100 PRI
TE (Cisco physical)	**NT** (NT physical)
2 (Rx+)	1 (Tx+)
1 (Rx-)	2 (Tx-)
5 (Tx+)	4 (Rx+)
4 (Tx-)	5 (Rx-)

Cisco PRI	Vega 400 PRI
NT (Cisco Physical)	**TE** (TE physical)
5 (Tx+)	1 (Rx+)
4 (Tx-)	2 (Rx-)
2 (Rx+)	4 (Tx+)
1 (Rx-)	5 (Tx-)

Cisco PRI	Vega 400 PRI
TE (Cisco Physical)	**NT** (NT physical)
2 (Rx+)	1 (Tx+)
1 (Rx-)	2 (Tx-)
5 (Tx+)	4 (Rx+)
4 (Tx-)	5 (Rx-)

Warning: Cisco pinouts are different to the Vega pinouts (note especially the .x+ and .x- swap-over) – you cannot unplug a cable that worked with a Cisco AS5300 and plug it into a Vega and expect it to work.

Annex 5 - Testing Vega and Cisco using NetMeeting

To narrow down issues and to verify your infrastructure it is always a good idea to use a less complex setup. This section explains how to make calls using Microsoft's NetMeeting. In this example we use NetMeeting 3.01 (4.4.3385) at IP address 10.10.11.2.
The configuration of NetMeeting is as follows:
Select a codec:

> Force NetMeeting to use a single codec:
> Tools→Options, Audio, Advanced, tick "Manually configure compression settings", select "CCITT, u-Law, 8000 kHz, 8 Bit, Mono"

NetMeeting and the Cisco AS5300

In NetMeeting, configure the Gateway:

> Tools→Options, Advanced calling, tick "use a gateway to telephones and videoconferencing systems", enter 10.10.11.203 (the AS5300).

- Configure a dial-peer on the AS5300:

	Command	Description
1.1	Router (config) # **dial-peer voice 3 voip**	Enter dial-peer configuration mode in order to create a dial-peer to handle calls to/from NetMeeting
1.2	Router (config-dial-peer) # **incoming called-number .** Router (config-dial-peer) # **destination-pattern 1099** Router (config-dial-peer) # **no modem passthrough** Router (config-dial-peer) # **voice-class codec 1** Router (config-dial-peer) # **session target ipv4:10.10.11.2** Router (config-dial-peer) # **dtmf-relay h245-signal** Router (config-dial-peer) # **no vad**	
1.3	Router (config-dial-peer) # **exit**	

You should now be able to make and receive calls with your PC using NetMeeting.

- Make an outbound call from NetMeeting: enter the phone number (e.g. 408 123-1234) and press the dial button. Accept the call and verify that you have two-way.
- Receive an incoming call on NetMeeting: Call (925) 555 1099, NetMeeting should ring. Accept the call and verify that you have two-way.

If either of these two tests does not succeed please contact Cisco's customer support (www.cisco.com).

NetMeeting and the Vega50 FXS

In NetMeeting, configure the Gateway:

Tools→Options, Advanced calling, tick "use a gateway to telephones and videoconferencing systems", enter 10.10.11.204 (the Vega50 FXS).

On the Vega50 FXS enter:

	Command	Description
1.1	ADMIN > **new plan**	
1.2	ADMIN > **set .planner.profile.1.plan.10.name=TestNetMeeting** ADMIN > **set .planner.profile.1.plan.10.srce=IF:.[^5],TEL:123** ADMIN > **set .planner.profile.1.plan.10.dest=IF:05,TEL:123,TA:10.10.11.2,CAPDESC:2**	
1.3	ADMIN > **set .h245.capdesc.2.caps=3**	Only offer G711ulaw codec

Some older versions of NetMeeting may require faststart to be turned off.

	Command	Description
2.1	ADMIN > **set .h323.use_fast_start=0**	Disable fast start for out going calls
2.2	ADMIN > **set .h323.accept_fast_start=0**	Disable fast start for incoming calls

After the test with NetMeeting restore the faststart configuration:

	Command	Description
2.1	ADMIN > **set .h323.use_fast_start=1**	Enable fast start for out going calls
2.2	ADMIN > **set .h323.accept_fast_start=1**	Enable fast start for incoming calls

Also NetMeeting does not like to be offered T.38 codecs – ensure that fax_codecs is configured to point to a set of voice only codecs (typically capability descriptor set 1) then no T.38 codecs will be offered.

	Command	Description
3.1	ADMIN > **set .h245.fax_capdesc.1.caps=1**	Sets preferred codec for fax to voice only codecs – i.e. NO fax codec

After the test with NetMeeting restore the fax descriptor configuration:

- Make an outbound call from NetMeeting: enter the phone number 5551001 and press the dial button. Accept the call and verify that you have two-way.
- Receive an incoming call on NetMeeting: Dial 123 from one of the phones attached to the Vega. NetMeeting should ring. Accept the call and verify that you have two-way.

Annex 6 - Basic Cisco configuration.

If the Cisco AS5300 is not already configured, certain elements will need to be set up. This section defines a basic configuration. Detailed testing and possibly further configuration will be required to ensure that the Cisco can make calls to the telephone network.

The example below configures an AS5300 with a T1 PRI trunk and an IP address of 10.10.11.203

Clear previous configs and set up password

Step 1	Command	Description
1.1	Router > **enable**	Log in
1.2	**Password: <password>**	Enter privileged mode
1.3	Router # **write erase**	Request configuration to be erased
1.4	… [CONFIRM] **y**	Confirm the erase
1.5	Power down AS5300 and re-power	
1.6	Would you like to enter initial configuration data [confirm yes/no] **no**	
1.7	Terminate auto install [yes] **yes**	
1.8	Router > **enable**	
1.9	Router # **configure terminal**	
1.10	Router (config) # **enable password <password>**	Set up password
1.11	Router (config) # **exit**	
1.12	Router # **exit**	

Log in, enter global setting section, enable IP routing and set up ip address

Step 2	Command	Description
2.1	Router > **enable**	Log in
2.2	**Password: <password>**	Enter privileged mode
2.3	Router # **configure terminal**	Enter configuration shell
2.4	Router (config) # **ip route 0.0.0.0 0.0.0.0. 10.10.11.1**	Set up static IP routing Ip route <network> <subnetmask> <next hop> [<distance>] Where <next hop> = default LAN gateway.
2.5	Router (config) # **hostname Router**	Set up hostname
2.6	Router (config) # **interface FastEthernet0**	
2.7	Router (config-if) # **ip address 10.10.11.203 255.255.255.0**	Configure IP address and subnet mask
2.8	Router (config-if) # **ip default-gateway 10.10.11.1**	Set up Lan gateway address
2.9	Router # **interface FastEthernet0**	
2.10	Router (config-if) # **duplex auto**	
2.11	Router (config-if) # **speed auto**	
2.12	Router (config-if) # **no shutdown**	Enable IP interface
2.13	Router (config-if) # **exit**	

Configure ISDN T1

Step 3	Command	Description
3.1	Router (config) # **isdn switch-type primary-5ess**	Configure trunk for 5ESS; Also it sets up the following default settings: • ip classless • no ip http server • mta receive maximum-recepients 0
3.2	Router (config) # **controller T1 0**	Set up the first T1 interface (index=0)
3.3	Router (config) # **no clock source line primary**	Configures Cisco as TE – to configure as NT, set: clock source line primary
3.4	Router (config) # **framing esf**	
3.5	Router (config) # **linecode b8zs**	
3.6	Router (config) # **pri-group timeslots 1-24**	T1 use timeslots 1-24
3.7	Router (config) # **description Pri Interface 0**	
3.8	Router (config) # **exit**	
3.9	Router (config) # **interface Serial0:23**	Set up a serial interface (D channel – channel 23) – keep the default values: • isdn switch_type primary-5ess • no ip address • no cdp enable • isdn T321 0
3.10	Router (config-if) # **no ip address**	
3.11	Router (config-if) # **isdn incoming-voice modem**	
3.12	Router (config-if) # **fair queue 64 256 0**	Fair queue <congestive discard threshold> <dynamic queues> <reservable queues>
3.13	Router (config-if) # **exit**	
3.14	Router (config) # **controller T1 1**	Set up the second T1 interface (index=1)
3.15	Router (config) # **no clock source line secondary**	Configures Cisco as TE – to configure as NT, set: clock source line secondary
3.16	Router (config) # **framing esf**	
3.17	Router (config) # **linecode b8zs**	
3.18	Router (config) # **pri-group timeslots 1-24**	T1 use timeslots 1-24
3.19	Router (config) # **description Pri Interface 1**	
3.20	Router (config) # **exit**	
3.21	Router (config) # **interface Serial1:23**	Set up the second serial interface (D channel – channel 23) – keep the default values: • isdn switch_type primary-5ess • no ip address • no cdp enable • isdn T321 0
3.22	Router (config-if) # **no ip address**	

3.23	Router (config-if) # **isdn incoming-voice modem**	
3.24	Router (config-if) # **fair queue 64 256 0**	
3.25	Router (config-if) # **exit**	

Optionally enable telnet access

Step 4	Command	Description
4.1	Router (config) # **line vty 0 4**	
4.2	Router (config-line) # **password 0 xxxx**	xxxx is the password for accessing the telnet interface
4.3	Router (config-line) # **login**	
4.4	Router (config-line) # **exit**	

Now configure the AS5300 and Vega as defined in the main part of this document.

Annex 7 - Enabling T.38 Fax

To use T.38 between the Vega and AS5300, the AS5300 must be running IOS >= 12.2(11) and the Vega must be running R4_T024 or greater firmware.

Additional configuration for the AS5300:

	Command	Description
	Router (config) # **voice call carrier capacity active**	
	Router (config) # **dial-peer voice 2 voip**	
	Router (config-dial-peer) # **fax rate 14400**	14.4kbps max Fax connect rate
	Router (config-dial-peer) # **fax protocol t38 ls-redundancy 0 hs-redundancy 0**	
	Router (config-dial-peer) # **no fax-relay ecm disable**	Enable fax relay (T.38) Error Correction Mode.
	Router (config-dial-peer) # **no vad**	
	Router (config-dial-peer) # **exit**	
	Router (config) # **voice service voip**	
	Router (config-voi-serv) # **fax protocol t38 ls-redundancy 0 hs-redundancy 0**	
	Router (config-voi-serv) # **h323**	
	Router (config-serv-h323) # **h245 caps mode restricted**	
	Router (config-serv-h323) # **exit**	
	Router (config-voi-serv) # **exit**	
	Router (config) # **fax interface-type fax-mail**	Alternatively fax interface type can be specified as vfc (instead of fax-mail) – Voice Fax Card.
	Router (config) # **voice-port 0:D**	
	Router (config-voiceport) # **no echo-cancel enable**	… for fax
	Router (config-voiceport) # **bearer-cap 3100hz**	Force ISDN bearer capability
	Router (config-voiceport) # **exit**	

Save and make active

Additional configuration for the Vega:

	Command	Description
	ADMIN > **set .h245.cap.6.name=t38udp**	Sets T.38 as capability 6
	ADMIN > **set .h245.capdesc.3.caps=6**	Capdesc 3 = t38udp only
	ADMIN > **set h245.fax_capdesc_index=3**	Sets preferred fax codec for t38udp not t38tcp)
	ADMIN > **set dsp.t38.tcf=transferred**	Sets Vega to handle modem negotiations across the UDP link (rather than doing them locally)
	ADMIN > **set _advanced.dsp.buffering.fax.enable=1**	Enables Vega to cope with Cisco sending out of sequence command (T.30) packets

Save and reboot

By default the Vega will only detect fax tones on calls it receives on its LAN interface (as required by the specification). If the Cisco does not detect and initiate T.38 fax transfer when calls are sent to it over the LAN, the Vega can be configured to detect and act upon fax calls arriving both on the LAN interface and the telephony interface. To do this, configure:

	Command	Description
	ADMIN > **set .sip.fax_detect=always**	Set Vega to detect and act upon fax tones received over its LAN and telephony interfaces

If command packets from the Cisco are missing or found to be transmitted by the Cisco gateway in a non-sequential order, try setting the ls-redundancy value > 0. This affects the data sent; it will send the current command data plus up to 5 previous sets of command data in each command message.

	Command	Description
	Router (config-dial-peer) # **fax protocol t38 ls-redundancy 5 hs-redundancy 0**	

Annex 8 - SIP configuration for Cisco 2691

	Command	Description
	Router (config) # **voice service voip**	
	Router (config-voi-serv) # **sip**	
	Router (config-serv-sip) # **exit**	
	Router (config-voi-serv) # **exit**	
	Router (config-serv-sip) # **access-list 101 permit ip any any**	
	Router (config-serv-sip) # **call rsvp-sync**	
	Router (config) # **dial-peer voice 100 voip**	
	Router (config-dial-peer) # **application session**	
	Router (config-dial-peer) # **destination-pattern 1.....**	
	Router (config-dial-peer) # **rtp payload-type nte 119**	
	Router (config-dial-peer) # **voice-class codec 1**	
	Router (config-dial-peer) # **voice-class sip url sip**	
	Router (config-dial-peer) # **session protocol sipv2**	
	Router (config-dial-peer) # **session target ipv4:x.x.x.x**	
	Router (config-dial-peer) # **dtmf-relay rtp-nte**	
	Router (config-dial-peer) # **fax rate 14400**	
	Router (config-dial-peer) # **fax protocol t38 ls-redundancy 0 hs-redundancy 0**	
	Router (config-dial-peer) # **no vad**	
	Router (config-dial-peer) #**exit**	
	Router (config) # **sip-ua**	
	Router (config-sip-ua) # **retry invite 5**	
	Router (config-sip-ua) # **timers trying 501**	
	Router (config-sip-ua) # **no oli**	
	Router (config-sip-ua) # **sip-server ipv4:x.x.x.x**	
	Router (config-sip-ua) # **no transport tcp**	Use UDP, not TCP to transport SIP messages
	Router (config-sip-ua) # **exit**	

Vega code must be >= R4 T029
Cisco code tested = Version 12.2(11)T2

Annex 9 - Prefix registration for Cisco Call Manager (H.323)

In H.323, as part of gatekeeper registration, a gateway registers the telephone number prefixes that it can handle for calls from VoIP to telephony. Vega gateways register the prefixes defined in their dial plans for dial plans whose source interface is IF:05 – prefixes are indicated by the dial plan telephone number ending with '.*'.

Cisco Call Manager however requires that each prefix entry is terminated by a # character. Extra dial plan entries are therefore required in the Vega to provide the prefixes to the Call Manager in the format it wishes to see them.

For example, for handling calls to telephony numbers starting with 404, 1344 and 506:

For routing calls, the three dial plan source expressions are:

```
IF:05,TEL:404.*
IF:05,TEL:1344.*
IF:05,TEL:506.*
```

To provide the Prefixes in Cisco Call Manager format, also add dial plans with source expressions:

```
IF:05,TEL:404#.*
IF:05,TEL:1344#.*
IF:05,TEL:506#.*
```

Contact Details
Email: support@vegastream.com
Web: www.vegastream.com
www.vegaassist.com

EMEA Office	USA Office
VegaStream Limited	VegaStream Inc.
The Western Centre	6200 Stoneridge Mall Road
Western Road	3rd Floor
Bracknell	Pleasanton
Berks RG12 1RW	California 94588
UK	USA
+44 (0) 1344 784900	+1 925 399 6428

