

Partner: Polycom
 Model: SoundStructure
 Device Type: DSP



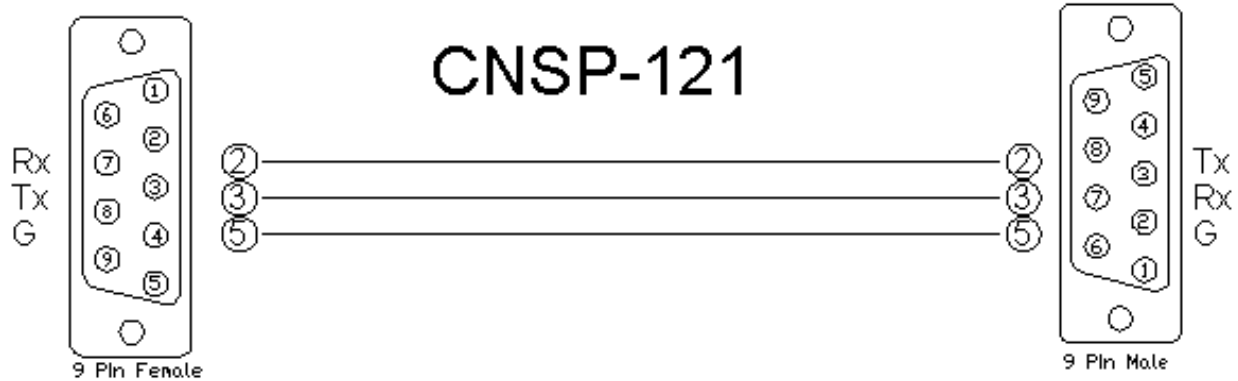
GENERAL INFORMATION

SIMPLWINDOWS NAME:	Polycom SoundStructure VoIP Phone Out Control v2.1
CATEGORY:	Conferencing
VERSION:	2.1
SUMMARY:	This module will control the telephony output on the VoIP card on the Polycom SoundStructure.
GENERAL NOTES:	<p>This module will control the telephony output on the Polycom SoundStructure VoIP card. The Virtual_Name_Out\$ must be connected to one of the Virtual_Name_In_*\$ inputs on the feedback processor module. The From_Processor\$ input must be connected from the corresponding To_Virtual_Name_*\$_Module\$ output. For this module to work, the Initialize input on the Feedback Processor module must be pulsed. Once the initialization is complete, the module should work and provide true feedback. The Fader inputs and outputs correspond to Fader controls for the input when using the Polycom SoundStructure software.</p> <p>This module supports a VoIP card configured with one or two lines. The VoIP card can be configured to use up to 12 lines so to support the maximum configuration; this module is designed to be modified by an experienced programmer.</p> <p>NOTE: POLYCOM RECOMMENDS END-USER VOLUME ADJUSTMENTS TO INPUTS AND OUTPUTS SHOULD BE PERFORMED WITH THE FADER CONTROLS AS SHOWN IN THE BASIC MODULE.</p> <p>INPUT AND OUTPUT GAIN CONTROLS FOUND ON THE ADVANCED MODULE SHOULD ONLY BE USED TO SET THE APPROPRIATE GAIN OR ATTENUATION REQUIRED TO BE COMPATIBLE WITH THE ELECTRICAL SIGNAL LEVELS COMING FROM ANY EXTERNAL EQUIPMENT (SUCH AS MICROPHONES, AMPLIFIERS, ETC.) THAT ARE CONNECTED TO THE SOUNDSTRUCTURE DEVICE. THE INPUT AND OUTPUT GAINS SHOULD BE SET DURING THE ROOM INSTALL PROCESS USING THE SOUNDSTRUCTURE STUDIO DESIGN SOFTWARE THAT IS SHIPPED WITH THE PRODUCT AND AVAILABLE FROM POLYCOM'S WEBSITE.</p> <p>END-USER ADJUSTMENTS OF THE INPUT AND OUTPUT GAIN SETTINGS AFTER INSTALLATION ARE NOT REQUIRED.</p>
CRESTRON HARDWARE REQUIRED:	C2I-COM6, C2COM-2/3, C2ENET-1/2, C2I-MC3-COM2-232, C2I-MC3ENET-1
SETUP OF CRESTRON HARDWARE:	<p>RS232</p> <p>Baud: 9600</p> <p>Parity: None</p> <p>Data Bits: 8</p> <p>Stop Bits: 1</p> <p>TCP/IP:</p> <p>Port: 52774</p>
VENDOR FIRMWARE:	1.6.0-Beta2
VENDOR SETUP:	The Polycom SoundStructure must have a program loaded. The virtual channel names from that program must be entered in the parameter fields on the control

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	modules.
CABLE DIAGRAM:	CNSP-121



CONTROL:

On/Off_Hook/Toggle	D	Pulse to take the phone on and off hook.
Flash	D	Pulse to flash the phone receiver.
Redial	D	Pulse to dial the last number. This will redial the last number the Polycom dialed, not the last number entered in the Crestron module.
Ignore	D	Pulse to ignore the current incoming call. The far end will continue to hear the call ring, however the local ringing will stop and the Ringing_Fb and Incoming_Call_Fb outputs on the VoIP Phone In Control module will subsequently both go low.
Reject	D	Pulse to reject the current incoming call. The local ringing will cease and the Ringing_Fb and Incoming_Call_Fb outputs on the VoIP Phone In Control module will subsequently both go low.
VoIP_Answer	D	Pulse to answer an incoming call.
VoIP_Hold/VoIP_Resume	D	Pulse to hold or resume the currently active call. The "hold" function is only available if the system is in an active call or conference. The "resume" function is only available if the system is in an active call or conference which is already on hold.
VoIP_Forward	D	Pulse to forward an incoming call. To forward a call, the user must then enter the phone number or extension where the call will be forwarded and pulse VoIP_Forward again.
VoIP_Transfer	D	Pulse to begin the process of transferring an active call. Then to transfer the call, enter the phone number or extension of where the call should be transferred to and press VoIP_Send to call that extension. The user will have the opportunity to speak to the transfer recipient (supervised transfer) prior to transferring the call, which is done by pulsing VoIP_Transfer again. Alternatively, if the call should be transferred without speaking to the person it is being transferred to, the "blind transfer" button can be pressed after "transfer" is pressed.

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		The "transfer" function is only available if the system is in an active call with a single party.
VoIP_Transfer_Blind	D	<p>Pulse to change an active transfer operation to a blind transfer. The call will immediately be transferred to another party without the opportunity to speak to that party first.</p> <p>The "transfer blind" function is only available if the "transfer" button has been pressed and a dial tone has been detected after the "transfer" button was pressed.</p>
VoIP_Conference	D	<p>Pulse to begin the process of starting a conference call. This can be used to either create a multiparty conference or to place call on hold while dialing another call using the same line.</p> <p>To create a conference call, pulse VoIP_Conference while in an active call. Once a dial tone is heard, begin dialing another phone number or extension and pulse VoIP_Send to connect to that extension. Once connected to the third party, pulsing VoIP_Conference again will join all three parties together into a single conference call.</p> <p>The "conference" function is only available if the system is in an active call.</p>
VoIP_Conference_Join	D	<p>Pulse to combine two or more separate calls into a single conference call where all of the remote parties can hear each other.</p> <p>The "join" function is only available if the system is in two or more separate active calls.</p>
VoIP_Conference_Split	D	<p>Pulse to break up an active conference call into multiple separate calls where the remote parties can no longer hear each other.</p> <p>The "split" function is only available if the system is in a conference call with two or more parties.</p>
VoIP_Send	D	<p>Pulse to immediately dial any entered digits. This is particularly useful when transferring a call or setting up a conference call. This function is optional as the system will automatically time out based on the dial plan when digits are entered, however this can be used to expedite the dialing process.</p> <p>The "send" function is only available if the system is in a state where dialing is permitted and digits have been entered.</p>
VoIP_Cancel	D	<p>Pulse to cancel an attempt to transfer a call or set up a conference call. The system will resume with the previously active call.</p> <p>The "cancel" function is only available when attempting to transfer a call or when setting up a conference call.</p>
VoIP_Do_Not_Disturb_On/Off/Toggle	D	Pulse to enable, disable, or toggle the Do Not Disturb state of the VoIP card. The DND state applies to the entire VoIP card, rather than an individual line or lines.
VoIP_Select_Line_*	D	<p>Pulse to select the corresponding line. This will also take the corresponding line off hook and a dial tone will be heard.</p> <p>A given line can only be selected if that line has been assigned a label.</p>
Select_Call_Appearance	A	Choose which call appearance's information should be displayed on the touch panel and which call appearance the controls on this module should be routed to. This function is for advanced users only and should not be needed for most systems. Valid values range from 1 to 24.
Number_In\$	S	Serial input for a phone number coming from an address book, Serial Send symbol or Serial I/O symbol. Only 0-9, *, # and comma (pause) may be used.

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Key_*	D	Pulse to enter the phone number to dial. When the phone is off hook, this will send the DTMF tones.
Dial	D	Pulse to send the entered phone number to the Polycom to dial.
Output_Fader_Up/Down	D	Press and hold to adjust the input fader.
Output_Fader_Value_In	A	Analog input to allow preset values to be sent to the Polycom SoundStructure. These values must be in the range set by the maximum and minimum parameters. They would generally come from a symbol like the Analog Initialize. These cannot come from a slider on a touch panel. Valid range is -100d to 20d. This input may be commented out if you do not wish to use presets from inside the Crestron.
Output_Mute_On/Off/Toggle	D	Pulse to mute the incoming signal from the phone.
From_Processor\$	S	Serial signal to be routed from a To_Virtual_*_Module\$ output on the feedback processor module.

PARAMETER:

Phone Out Name	P	Name entered in the Polycom SoundStructure software for the phone input you want to control.
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FEEDBACK:

On/Off_Hook_Fb	D	High to indicate the current hook state.
VoIP_Hold_Visible_Fb	D	<p>High to indicate the "hold" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user.</p> <p>The "hold" function is only available if the system is in an active call or conference which is not already on hold.</p>
VoIP_Resume_Visible_Fb	D	<p>High to indicate the "resume" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user.</p> <p>The "resume" function is only available if the system is in an active call or conference which is already on hold.</p>
VoIP_Transfer_Visible_Fb	D	<p>High to indicate the "transfer" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user.</p> <p>The "transfer" function is only available if the system is in an active call with a single party.</p>
VoIP_Transfer_Blind_Visible_Fb	D	<p>High to indicate the "transfer blind" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user.</p> <p>The "transfer blind" function is only available if the "transfer" button has been pressed and a dial tone has been detected after the "transfer" button was pressed.</p>

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VoIP_Conference_Visible_Fb	D	High to indicate the "conference" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user. The "conference" function is only available if the system is in an active call.
VoIP_Conference_Join_Visible_Fb	D	High to indicate the "join" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user. The "join" function is only available if the system is in two or more separate active calls.
VoIP_Conference_Split_Visible_Fb	D	High to indicate the "split" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user. The "split" function is only available if the system is in a conference call with two or more parties.
VoIP_Send_Visible_Fb	D	High to indicate the "send" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user. The "send" function is only available if the system is in a state where dialing is permitted and digits have been entered.
VoIP_Cancel_Visible_Fb	D	High to indicate the "cancel" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user. The "cancel" function is only available when attempting to transfer a call or when setting up a conference call.
VoIP_Do_Not_Disturb_On/Off_Fb	D	High to indicate that Do Not Disturb is currently enabled or disabled. DND applies to the entire VoIP card.
VoIP_Line_*_Selected_Fb	D	High to indicate the current line is selected. This output will change state automatically based on which line is currently in use on the VoIP card.
VoIP_Select_Line_*_Visible_Fb	D	High to indicate the corresponding "line select" button should be visible on the GUI. This is an optional feature that allows the programmer to create a cleaner user interface where buttons that cannot be used in the current state of the system are hidden from the user. A line select button is only available if the line is assigned a label.
VoIP_Line_*_Label\$	S	Reports the label assigned to each line. This is typically the phone number or extension for each line.
Call_Appearance_Fb	A	Indicates which numeric ID of currently active call appearance. Values range from 1 to 24.

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VoIP_Info_Field*\$	S	Two lines of text intended to be displayed on a touch panel or user interface. These lines mimic the two lines of text found on a Polycom VoIP phone.
VoIP_Call_State\$	S	Reports the state of the current call appearance as serial text. Can be tied to a Serial I/O for triggering logic related to a specific state.
Number\$	S	Serial signal indicating the phone number entered.
Output_Fader_Bar	A	Analog value indicating the current fader value. To be displayed using a bar graph on a touch panel.
Output_Mute_On/Off_Fb	D	High to indicate the current mute state.
Virtual_Name_Out\$	S	Serial signal to be routed to one of the Virtual_Name_In_*\$ inputs on the feedback processor module.
To_Device\$	S	Serial signal to be routed to a 2-way serial com port or a TCP/IP Client.

TESTING:

OPS USED FOR TESTING:	PRO2: 4.003.0015 MC3: 1.003.0008
SIMPL WINDOWS USED FOR TESTING:	4.03.20
DEVICE DB USED FOR TESTING:	73.05.002.00
CRES DB USED FOR TESTING:	55.00.011.00
SYMBOL LIBRARY USED FOR TESTING:	985
SAMPLE PROGRAM:	Polycom SoundStructure v2.1 Demo
REVISION HISTORY:	<p>V. 1.0 – Original Release</p> <p>V. 1.1 – Added level maximum and minimum feedback from the Polycom. These values are used to adjust the scale of the bar graph and prevent ramping beyond these points. Removed the maximum and minimum parameter fields on the module.</p> <p>V. 1.2 – Created separate basic and advanced controls. The basic controls do not offer control of the Level. Polycom recommends using the basic modules to control the SoundStructure.</p> <p>V. 2.0 – Now 3-Series compatible. Improved performance by reducing extraneous find() calls. Added logic to handle virtual names containing double quote characters by escaping them with a backslash.</p> <p>V. 2.1 – Incorporated 3-series best practices as needed in all Simpl+.</p>