GLOBALCOM®.IP

Topic: Telephony with GCK

Terms and Abbreviations Used in This Document

BRI – Basic Rate Interface.

CODEC – Coder / Decoder. An encoding method for digital media.

DTMF – Dual Tone Multi Frequency: Key presses on a phone.

G.711 μ-Law – an 8 kbit/s audio CODEC providing approximately 3 kHz of audio bandwidth. μ-Law is commonly used in North America and Japan.

G.711 A-Law – an 8 kbit/s audio CODEC providing approximately 3 kHz of audio bandwidth. A-Law is commonly in non- μ-Law locations or when one leg of the call terminates in an A-Law location.

G.722 – a 16 kbit/s audio CODEC providing approximately 7kHz of audio bandwidth. Requires audio infrastructure capable of wideband audio.

H.323 – An ITU standard for VoIP communications.

ISDN – Integrated Services Digital Network.

ITU – International Telecommunications Union.

POTS – Plain Old Telephone System.

PRI – Primary Rate Interface.

PSTN – Publicly Switched Telephone System.

RTP – Real-time Transport Protocol. Used for media and DTMF. This is an open standard defined in RTF 3550 and subsequent related documents.

SIP – Session Initiation Document. Used for control and signaling. This is an open standard defined in RFC 3261 and subsequent related documents.

VoIP – Voice over Internet Protocol. A methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as corporate networks and the Internet.

Overview

GCK contains rich telephony features. GCK leverages open standards such as SIP and RTP to take user entry and route audio using many different types of phones and phone systems. This document describes a list of features in GCK, followed by a series of Case Studies detailing how to use GCK to accomplish both simple and complex workflow needs. Internally, GCK can be configured to implement an array of Extensions, which are softphones capable of routing telephone audio or initiating announcements. These capabilities are detailed below.

Compatibility

GCK can interface to any phone system capable of communicating using SIP and RTP. GCK has successfully connected to all major versions of VoIP including Cisco, Avaya (IP Office and Call Manager), Shortel, less common systems such as 3CX, Asterisk, and FreeSWITCH, even hosted solutions such as Comporium and Unity. Other telephony systems (such as hybrids or H.323) may not natively support SIP; however, SIP compatibility modules may be available to add SIP functionality to them.

GCK uses the universal G.711 CODEC (either µ-Law or A-Law) for low-bandwidth telephony, or the wideband G.722 CODEC when available for higher quality audio.

Features

Connection Modes

GCK as a Series of Extensions
In Extensions Mode, GCK can interface to a VoIP PBX as a series of extensions. To the PBX, these appear as any other Basic SIP endpoint (phone). This provides simplicity of configuration. However several limitations may be present, such as a limited number of extensions supported by the PBX due to technical or licensing limitations.

GCK as a Trunk
In Trunking Mode, GCK can appear to the PBX as a trunk. A trunk is a “peer” connection as an external SIP system. This is the preferred method of integration. In this manner, GCK can have a “functionally-infinite” number of extensions.

GCK as its Own PBX
In Internal Only Mode, GCK can take command of its own VoIP phones. This is useful when in small or self-contained systems. No “House PBX” is required.

Non-VoIP Telephone Communications
GCK can utilize analog phone lines to communicate with legacy non-VoIP PBXs or with the PSTN. This is accomplished with an FXO gateway, which converts analog calls to the VoIP calls required by GCK. An FXO Gateway allows connection to the PSTN. A customer may chose to connect via analog lines to simplify internal Ethernet network configuration to segregate the GCK to a private network. GCK can also use other gateway devices to generate SIP connections from other telephony networks such as SS7, PRI, BRI, or ISDN. Conveniently, these gateway devices appear to GCK as a trunk, and are configured exactly the same way. GCK has successfully been connected to gateways from Cisco, Grandstream, MultiVoIP, and others. AtlasIED offers support for the Grandstream GXW-4104 or GXW-4108 Gateways.

Security
GCK can be configured to allow calls based on PIN or admit calls based on CallerID, or allow both. This allows you to develop a tiered approach to user functionality, while allowing the obvious benefit of rejecting unauthorized user activity- especially when attached to the PSTN.

Types of Extensions

Direct Extensions
Direct Extensions perform a single action immediately upon answer, without user input. An action could be one of the following- zone page (live or delayed), zone group page (live or delayed), intercom connection, monitor a zone, play a message, or launch an event. This action will require no selection. Examples of such an extension would be a live page to all-call, initiate emergency evacuation notice, or play a shift-change horn or school bell. The GCK administrator can configure the action with the desired parameters such as priority, zone group number, audio take(s), etc.

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Prompting Extensions
Prompting Extensions implement a simple yet powerful IVR templating engine. GCK comes stock with 3 templates to cover many possibilities. However, the administrator can edit them or add new ones as required. A template can be assigned to an extension, or the extension may be configured to select a template based on the user chosen through the PIN or CallerID as described above. Audio takes for the templates may be uploaded, allowing them to be fully customized. You can make templates be language-specific or even upload special tones which indicate the user should enter certain values. GCK can accept any WAV file as a template prompt or entry prompt. It will resample the audio or mix the channels in the file as needed.

Templates offer a two-tier menu system, where the first level selects the type (e.g. Live zone page), and the second tier selects the destination (e.g. Zone number, event Id, classroom number). In addition, if a single item is in the template, the prompt starts immediately with the destination. Template nodes may be mixed with selection and a Direct Extension style node which does not require entering a destination.
Intercom, Monitoring, and Dialout
GCK can be used with CobraNet or IP Multicast intercom speaker devices when two-way audio is desired. In a classroom setting, a button can be configured to dial out of GCK to the house PBX (or PSTN). The intercom device description field is used as the CallerID to indicate which room is calling.

GCK Dialout is powered by sophisticated hunt groups, which easily allow complex dialing rules. If you want many phones to ring at once, a second phone to ring if the first one does not answer (or a third, or fourth, or…), this is easy to accomplish. For example, you could have several desk phones ring at once, then failover to a travelling wireless phone if no answer, perhaps with a final fallback to a cell phone.

In many intercom installations, the acoustics are not favorable for audio clarity- tile floors, square-ish room dimensions, etc. GCK contains a built in echo canceller capable of suppressing echo from non AEC-equipped devices.

Case Studies
The following fictional case studies illustrate a subset of the wide range of simple and complex use cases GCK can handle.

Case 1: An Eight Floor Hospital.
The facility is an eight floor hospital. They have Cisco Unified Call Manager (CUCM) installed. There is a phone at the nurse's station on each floor, as well as in various other departments. The hospital desires paging audio in all public spaces. They want an emergency all-call to all floors for code blue pages.

Solution
1. A single GCK IP-108 controller for the facility.
2. A DNA amplifier dedicated for each floor.
3. CUCM trunk to GCK.
4. Network routing from the voice VLAN to the paging VLAN.
5. A zone group, numbered 1 through 8 such that zone group 1 contains all the DNA zones for floor 1 … zone group 8 contains the zones connected to speakers on floor 8.
6. Ten Direct Action Extensions (x100 to x800) each configured to do a live page to the corresponding zone group (i.e. Extension x100 does a live page to zonegroup 1… Extension x800 does a live page to zonegroup 8.
7. A dialing rule in CUCM which directs the phone call to GCK. For example, if the nurse dials 7xxx, CUCM will know that '7' means a page, and the other numbers are the extension to call in GCK. So assuming the nurse dials 7100, GCK's extension x100 will answer. Audio will be routed to zonegroup 1. Likewise for 7200 – 7800.
8. An All Call zonegroup (100) which contains all zones.
9. A "Code Blue" Announcement Class with a higher priority than normal paging.
10. A Direct Extension (x911) which is configured to do a live page to zonegroup 100, using Announcement Class "Code Blue".Now if a nurse dials x7911, Direct Extension 911 goes live, pre-empting other announcements so that the important Code Blue is heard on all speakers.

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Case 2: A school With IP Speakers and several phones.
The facility is a middle school with 68 classroom spaces, four hallways, and a gym. They are connected to a hosted VoIP provider. In addition to normal bell scheduling, they want to be able to perform the following:

- Initiate an intercom to an individual room, or allow a room to initiate an intercom request.
- Page All Call for morning announcements.
- Initiate an emergency lockdown or stop an emergency lockdown to all-call.
- Allow the principal to monitor a room.
- Ring a "manual bell". Only the receptionist's phone will ring a manual bell.

Solution

1. Put an IP speaker in each room, including the library. Set the name of each speaker device appropriately, so that the CallerID at the receptionist's desk for a classroom-initiated intercom (known as a Dialout) appears as the caller on the destination phone.
2. Create an all-call zone group for all IP speaker zones.
3. Create the following Receptionist template:
   a. For Intercom Press one → Enter room number → intercom action to chosen room.
   b. For all-call Press two → Live all-call page.
   c. For manual bell Press 3 → Play manual bell PDRP.
4. Create the following Principal template:
   a. For Intercom Press one → Enter room number → intercom action to chosen room.
   b. For all-call Press two → Live all-call page.
   c. To monitor a room, Press three → Enter room number → monitor action to chosen room.
5. Create a prompting extension with the template set to "Template based on password". Check the Locked button.
6. Create a mic password for the receptionist. Set the CallerID to the receptionist's phone number. Set the default template to the Receptionist Template. Check Accept Phone Login.
7. Create a mic password for the principal. Set the CallerID to the principal's phone number. Set the default template to the Principal Template. Check Accept Phone Login.

Case 3: A train station with CUCM and Analog Failover.
A transit system uses Cisco Unified Call Manager. They want to page each station from the head end using their VoIP phone. As a backup, they want to use the PSTN in case the network or VoIP phone system is down.

Solution

1. An IP-108 at each station with a Grandstream GXW-4104 connected to the PSTN.
2. An IP-108 at the head end with a trunk to CUCM.
3. The controllers at each station are connected to the head end as remote zones.

Case 4: A small industrial plant with paging.
A manufacturing facility wants to have shift change bells, live paging to warehouse, production and office areas, and emergency evacuation. Their facility uses a non-VoIP PBX. They want one line to prompt for the zone using a simple tone of their choosing.

Solution

1. An IP-108 with DNA zones in the warehouse, production and office areas.
2. The shift change bells are PDRPs played as scheduled announcements.
3. The Emergency Evacuation message is a PDRP set to an Announcement Class with a higher priority than the live paging Announcement Class.
4. The house PBX (like most all) is capable of adding analog lines. Add two lines. Connect them to a Grandstream GXW-4104.
5. The IP-108 has a trunk connection to the Grandstream GXW-4104.
6. Upload the customer prompt as a telephone prompt. Label accordingly.
7. Creating the following template:
   a. Add a node, but do not put a key or prompt WAV.
   b. Change the Zone entry prompt to the prompt uploaded in Step 6.
8. Create a prompting extension. Set the template to the new template created in Step 7.
9. Create a Direct Extension. Set the action to play the Emergency Evacuation PDRP to All Call.

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Case 5: Multi lingual templates.
A major airport has agents who speak several languages. They wish to be prompted in their preferred language.

Solution 1

1. Create a template with prompts in language 1. Call it Lang1Template.
2. Create a template with the same prompts in language 2. Call it Lang2Template.
3. Create a prompting extension for language 1.
4. Set the template to Lang1Template.
5. Create a prompting extension for language 2.
6. Set the template to Lang2Template.
7. Tell users of language 1 to call extension 1 and users of language 2 to call extension 2.

Solution 2

1. Create a template with prompts in language 1. Call it Lang1Template.
2. Create a template with the same prompts in language 2. Call it Lang2Template.
3. Create a prompting extension with the template set to "Template based on password". Check the Locked checkbox.
4. Create a Mic Password for language1. Set the Base Template for this password to Lang1Template.
5. Create a Mic Password for language2. Set the Base Template for this password to Lang2Template.
6. Instruct users of language 1 to use PIN 1.
7. Instruct users of language 2 to use PIN 2.

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