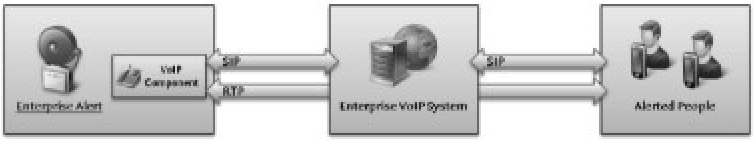
4.4.2 Setting up Voice Calls

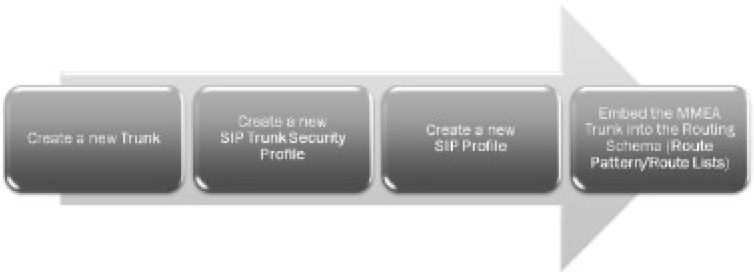
Enterprise Alert® realizes voice calls through a custom SIP based VoIP integration. This allows for great flexibility in your voice call scenarios, ranging from Cisco Unified Call Manager (CUCM) integration to complete replacements of outdated TAPI integrations. The Enterprise Alert® VoIP component can act as SIP Endpoint or as SIP Trunk.

The VoIP component has the following characteristics.

- Signaling is done according to the SIP RFC3261 (does not support H.232), supporting the "early offer" call/answer model.
- Media Sessions are negotiated using SDP (RFC2327) and transport is realized using RTP (RFC3550)
- Security SIPS or SRTP is currently not supported
- DTMF supports SIP-INFO requests (RFC2976) and RTP Payload for DTMF (RFC4733)
- Supported audio codec is G711 aLaw or uLaw (WAV file support but no PCM)



Before you configure a VoIP connection in Enterprise Alert you have to prepare your VoIP system i.e. create SIP Profiles and trunks. The preparation steps needed greatly depend on your VoIP system. A sample preparation workflow for CUCM would be:



Once your VoIP system is prepared, open *System > Notification Channels* and select VoIP as the media type in the Web portal of Enterprise Alert® to configure a VoIP connection. Configuration will look as below:

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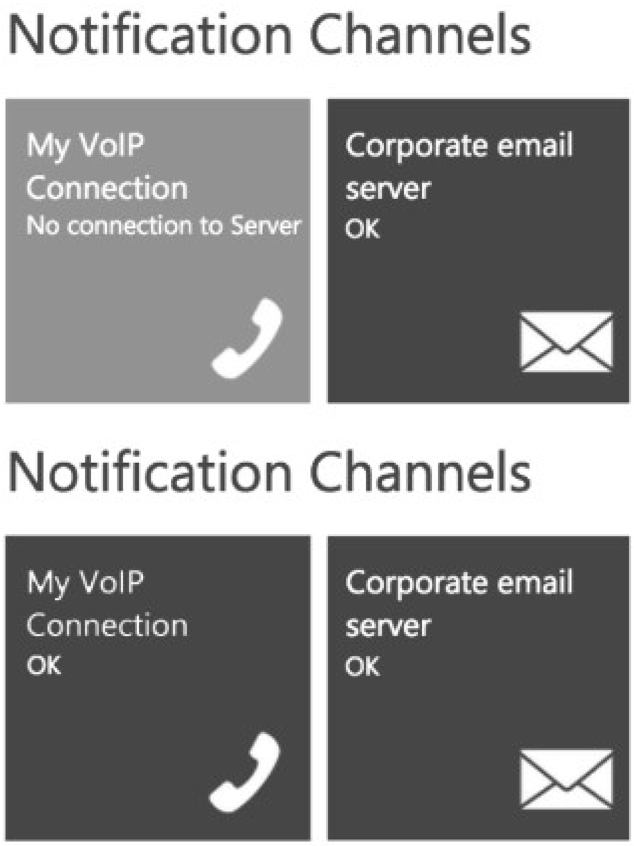
The following settings are required to set up a VoIP connection:

- Connection name: A unique name identifying the connection in Enterprise Alert®
- VoIP Server: IP address of your VoIP server
- Account Credentials: SIP URI or an account name and password if required for authentication on the VoIP server
- Connection Type: SIP Trunk or SIP Endpoint
- TTS Engine: The TTS engine which will be used for voice calls. TTS engines are gathered through .NET and UCMA interfaces. Legacy and nonstandard TTS engines may not be available. The TTS engine will be automatically switched to the first matching engine if the selected TTS engine does not support the current alert's language.

Other optional settings allow you reserve or disable inbound lines, reduce the amount of concurrent calls and fine tune call and input timeouts.



If the connection has been configured correctly, the new VoIP connection will appear dark-blue in the Notification Panels page, with 'OK' displayed in the bottom right corner. Otherwise it will appear orange with a short error message along the bottom of the tile.

> Creating the first VoIP connection will automatically create a message route directing ALL voice-calls through this connection. You should modify the message routes if you also have active Lync integrations to prevent failed submission of messages through the wrong connections. Setting the