



# Introduction to N2SRP INAP-Controlled IVR

---

N-SQUARED SOFTWARE

# What is N2SRP?

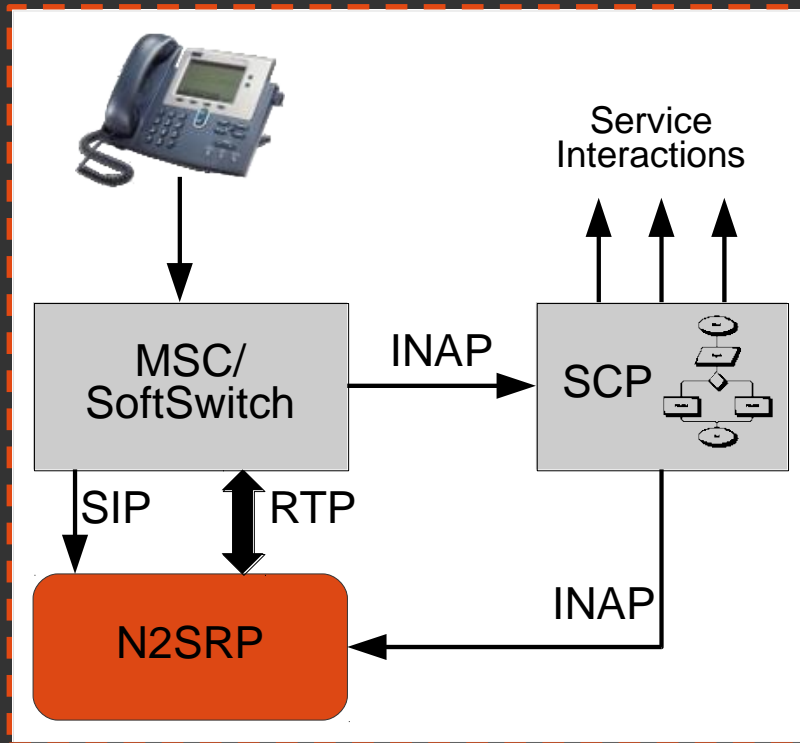
---

- N2SRP is an INAP-Controlled SIP IVR platform...  
...a “**Specialized Resource Platform**” (ITU-T Q.1200)
- The key features of an SRP are:
  - Voice Channels are controlled using **SIP** to a core network soft-switch.
  - Voice Channel Audio is carried over **RTP** to a core network interface.
  - Voice Interaction controlled by **INAP** or **CAP** from an **Intelligent Network SCP**.

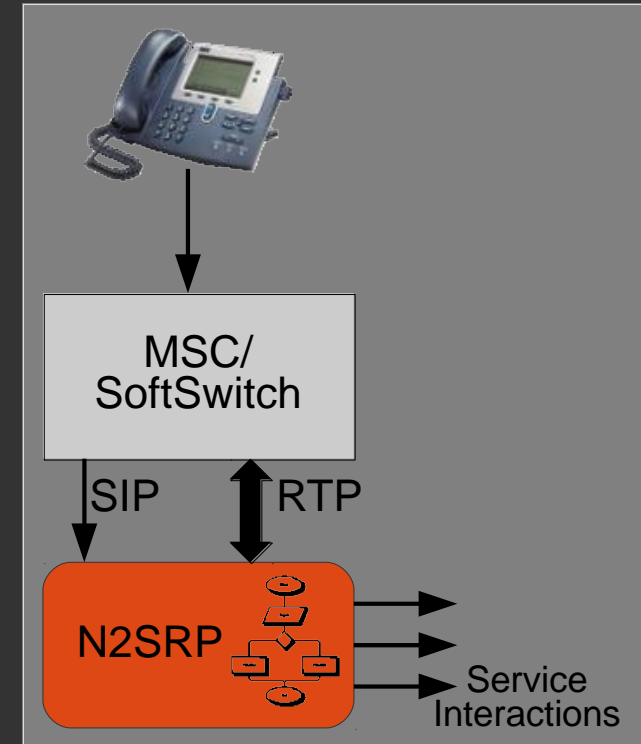
In the SRP model, Voice **Service Logic** is executed on the **SCP**, where it is centralized and consistent across voice and non-voice channels (e.g. SMS, USSD, Data, Diameter, SOAP, XML, etc.)

# N2SRP (An INAP-Controlled SIP IVR)

This Presentation describes **N2SRP**  
-- our **INAP-Controlled SIP IVR** platform.



This Presentation does not describe N2IVR  
-- our Standalone SIP IVR platform.



# When to use a SRP?

---

- An SRP is used when:
  - Implementing voice services which are controlled by an **INAP SCP**.
  - When the voice service needs to **Play Announcements**.
  - When the voice service needs to **Prompt for DTMF Input**.
- Typical Services using an SRP are:
  - Announcements & Menus for **Toll-Free, Premium, UAN, Tele-Voting**.
  - Service Announcements for IN-Controlled **Pre-Paid & Calling Card**.
  - **Self-Management** for IN Services – *low balance, balance query, friends & family, CUG, product type swap, voucher redeem, credit-card top-up, etc.*

# Features of the N-Squared N2SRP

---

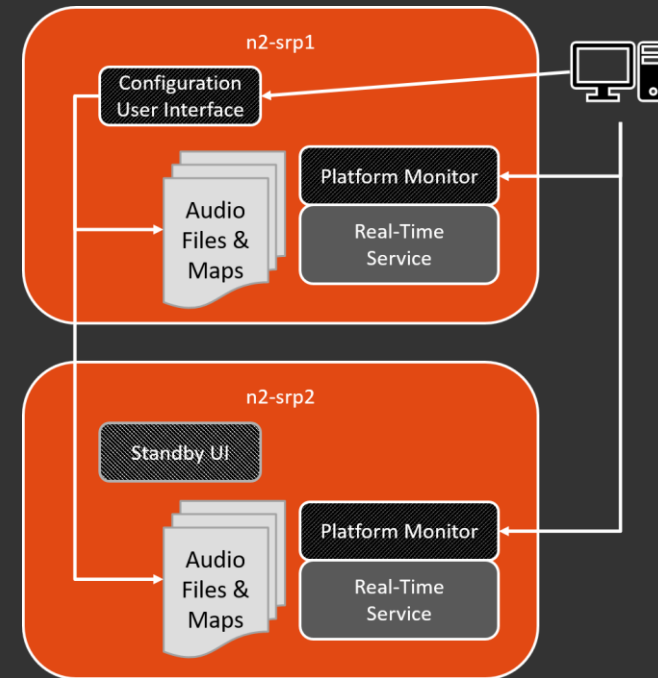
- **High-Availability** (N+1 Redundancy).
- **Linear Scalability** for additional capacity and geo-distribution.
- Generic **x86-64** Hardware (**Virtualized** or **Bare Metal**).
- Pluggable support for additional **Languages**.
- Cost-Effective **SIP/RTP** trunking.
- Control via:
  - **INAP/CAP** over **TCAP**, **SIGTRAN** (M3UA+SCCP or SUA), or
  - Option for Standalone Logic for non-INAP service implementation.

# Deployment

---

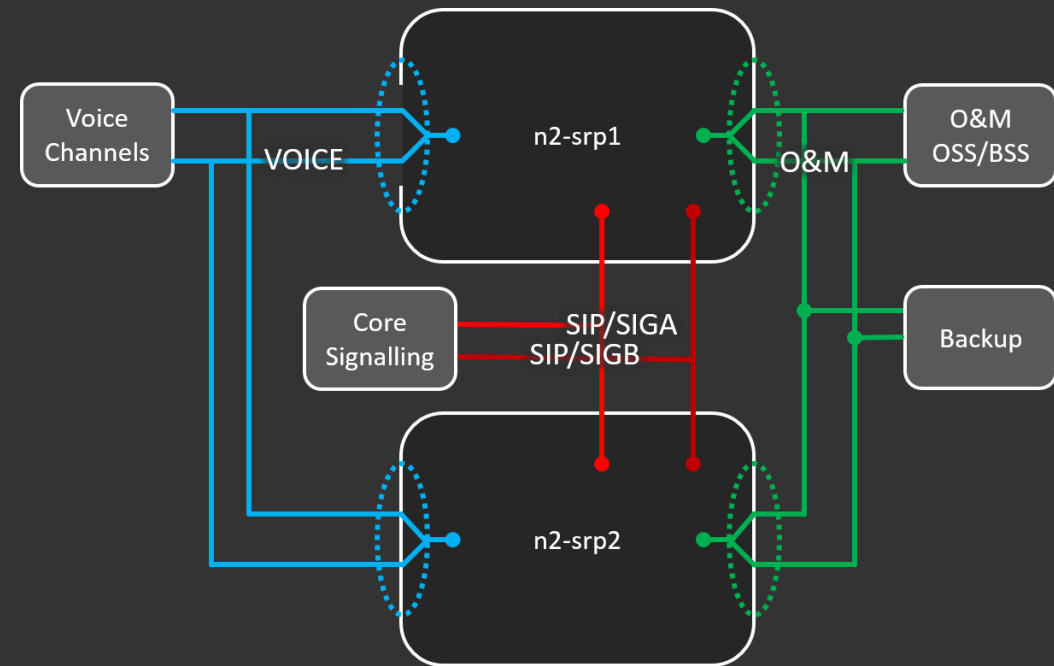
# Example Redundant Platforms

- Redundant Hosts for HA
  - N+1 Active/Active
  - Linear Scalable
- Single Point of Configuration
  - Convenient & Consistent
  - Secondary nodes are Standby
- Direct Real-Time Monitoring
  - Maximum Responsiveness
  - Maximum Resilience



# Example Redundant Network

- **Voice** bonded network for Real-Time RTP Audio communication.
- **Operations & Management** bonded network for Admin, Announcement Management, Monitoring, and Backup.
- SIGTRAN **primary** and **secondary** connections to Core Network, and SIP Session Management.





# Interfaces

---

## Network Integration

1. SIP to MSC (RFC 3261)
2. RTP to MSC (RFC 3550)
3. INAP to SCP (ETS 300 374-1)
  - SCTP (RFC 2960)
  - M3UA (RFC 4666), or
  - SUA (RFC 3868)
  - TCAP (ITU-T Q.771-775)

## OSS & BSS Integration

- A. Alarm via SNMPv2 (RFC 3416)
- B. Stats via Etsy StatsD
- C. Platform Admin via HTTP/S
- D. Configuration GUI, HTTP/S
- E. Configuration API, RESTful over HTTP/S

Note: Interfaces are implemented to the extent necessary to support advertised features.  
Refer to the product Protocol Conformance Statement documentation for details.

# User Interfaces

---

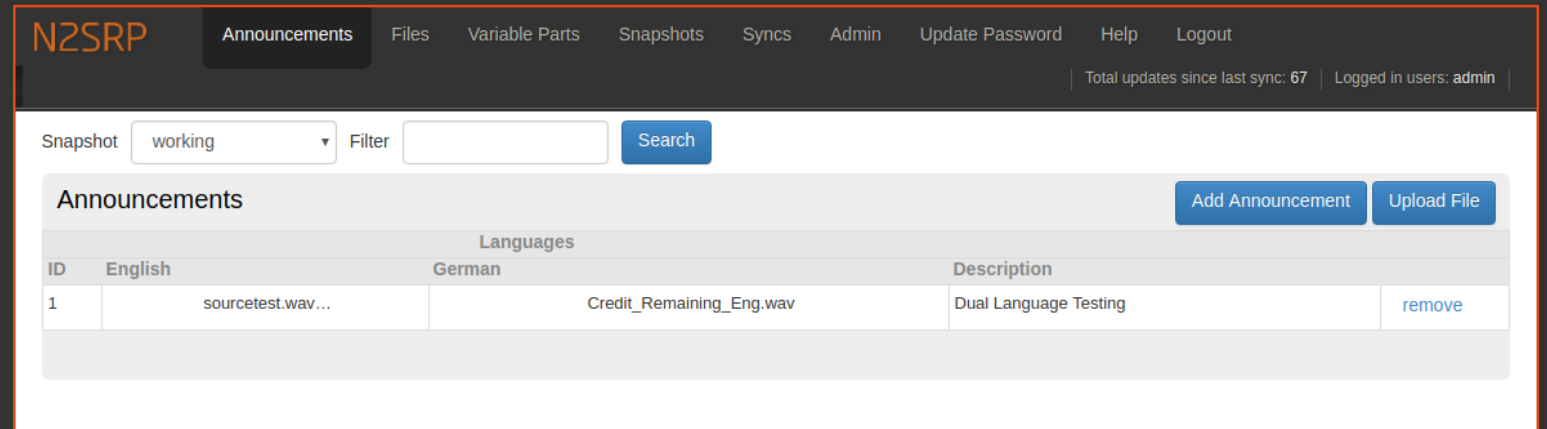
# Configuration User Interface

The Configuration User Interface manages audio files, and maps the Announcement IDs used by the INAP SCP for controlling interactions.

The interface is pure-web with support for all modern browsers and tablets (without Java/Flash/ActiveX/Citrix).

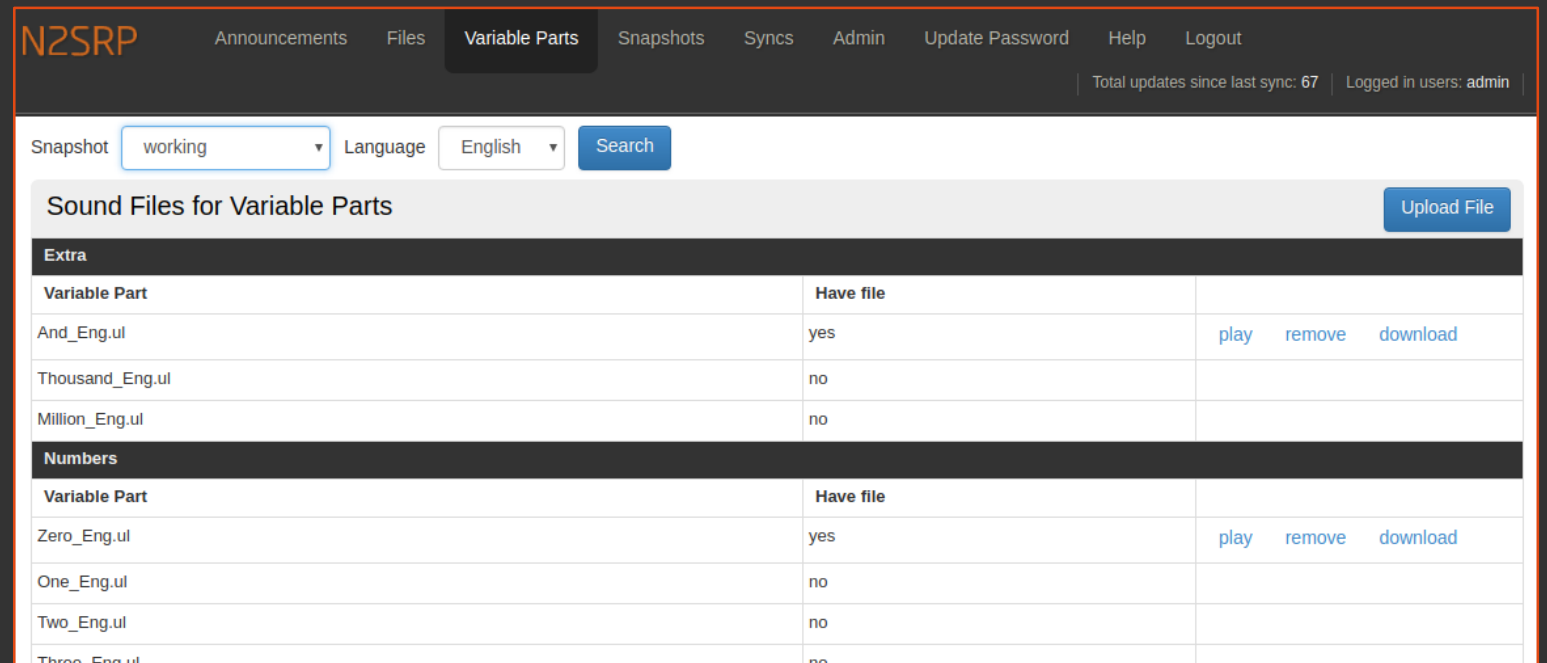
Audio files can be uploaded, downloaded, or played from within the browser.

The UI performs automatic audio file-type detection and conversion.



The screenshot shows the 'N2SRP' web interface with the 'Announcements' tab selected. The top navigation bar includes links for Announcements, Files, Variable Parts, Snapshots, Syncs, Admin, Update Password, Help, and Logout. A status bar indicates 'Total updates since last sync: 67' and 'Logged in users: admin'. Below the navigation, there's a 'Snapshot' dropdown set to 'working' and a 'Filter' input field. The main content area is titled 'Announcements' and features a table with columns for ID, English, German, and Description. A 'remove' link is present for each entry. Buttons for 'Add Announcement' and 'Upload File' are located at the top right of the table.

ID	Languages		Description	
	English	German		
1	sourcetest.wav...	Credit_Remaining_Eng.wav	Dual Language Testing	<a href="#">remove</a>



The screenshot shows the 'N2SRP' web interface with the 'Variable Parts' tab selected. The top navigation bar is the same as the previous screenshot. The status bar also shows 'Total updates since last sync: 67' and 'Logged in users: admin'. Below the navigation, there's a 'Snapshot' dropdown set to 'working' and a 'Language' dropdown set to 'English'. The main content area is titled 'Sound Files for Variable Parts' and features a table with columns for Variable Part, Have file, and actions (play, remove, download). The table is divided into two sections: 'Extra' and 'Numbers'.

Variable Part	Have file	
And_Eng.ul	yes	<a href="#">play</a> <a href="#">remove</a> <a href="#">download</a>
Thousand_Eng.ul	no	
Million_Eng.ul	no	

Variable Part	Have file	
Zero_Eng.ul	yes	<a href="#">play</a> <a href="#">remove</a> <a href="#">download</a>
One_Eng.ul	no	
Two_Eng.ul	no	
Three_Eng.ul	no	

# Configuration UI (cont.)

The N2SRP supports fine-grained security, as required for MVNO/MVNE sites where a single hardware SRP platform is shared by multiple co-resident operators.

This can also be relevant where Testing and Production share a single SRP platform, or when the announcement platform is connected to multiple different Intelligent Networking environments.

N2SRP

[Announcements](#)[Files](#)[Variable Parts](#)[Snapshots](#)[Syncs](#)[Admin](#)[Update Password](#)[Help](#)[Logout](#)

Total updates since last sync: 67 | Logged in users: admin

Announcement ID Groups

Name	Min Announcement ID	Max Announcement ID		
Second Group	4	151	<a href="#">edit</a>	<a href="#">remove</a>
Testing Group	100	202	<a href="#">edit</a>	<a href="#">remove</a>

Add Group

Users

Username	Access Level	Announcement ID Group		
admin	Administrator		<a href="#">edit</a>	<a href="#">remove</a>
Group User	Restricted User	Testing Group: [100 → 202]	<a href="#">edit</a>	<a href="#">remove</a>
Another User	Restricted User	Testing Group: [100 → 202]	<a href="#">edit</a>	<a href="#">remove</a>
test	Restricted User	Second Group: [4 → 151]	<a href="#">edit</a>	<a href="#">remove</a>
nsquared	Restricted User	Second Group: [4 → 151]	<a href="#">edit</a>	<a href="#">remove</a>

Add User

SRP Servers

Hostname/IP Address	Username	SSH Identity File	Comments
---------------------	----------	-------------------	----------

Add SRP

# Monitoring User Interface

The run-time service execution environment for N2SRP offers an HTTP/S port to perform monitoring and system administration activities using any modern web browser.

The interface allows access to:

- ☐ In-Progress Call Instances.
- ☐ In-Progress SIP Txns & Dialogs.
- ☐ Working Configuration.
- ☐ Current Statistics.
- ☐ Trace Logs for Call Instances.

SRP AVAILABLE	0=# Queued Timeouts 1=# <u>Instances (Active)</u> 1=# <u>Instances (Active/Retained)</u> 1=# <u>EDR Files (Recent)</u> 0=# EDR #Pending 1=# <u>TCAP TIDs</u> 1=# <u>SIP Servers</u> 0=# <u>SIP Transactions</u> 1=# <u>SIP Dialogs</u> 2=# <u>RTP Workers</u> 3=# <u>Languages</u> 1=# <u>SCP Addresses</u>	2=RTP.workers.count 1=TCAP.ARI 1=TCAP.PACUI 1=instance.created 1=instance.start 1=requests.ACK 1=requests.INVITE	Trace Level = 1 ▾ Instance Retention = 50 Enable EDRs = YES EDR Directory = /tmp/edr EDR Prefix = n2srp EDR Hostname = vagabond EDR #Files Shown = 20 EDR Retry Interval = 30 EDR Report Interval = 5 EDR Max EDR/File = 2000 EDR Max secs/File = 300 EDR Max EDR/Flush = 20 EDR Max secs/Flush = 5 SIP Public Host = vagabond SIP Public Port = 5060 SIP Bind Host = 0.0.0.0 SIP Bind Port = 5060 SIP EDR Merge = YES ▾ RTP Pool Size = 2 RTP Local IP = 10.42.2.251 RTP Start Port = 6970 RTP Event SendOnly = 0 Use Early Media = NO DTMF Detection = local DTMF End Digit = # DTMF Cancel Digit = * DTMF Min Digits = 1 DTMF Max Digits = 31 DTMF Cancel Digit = * DTMF First Digit T/0 = 8 DTMF Inter Digit T/0 = 4 DTMF Interruptable = 1 Leading Silence (ms) = 40 SIGTRAN App = SIGTRAN SCP ID Location = after SCP ID Len = 1 Correlation ID Len = 3 INAP Variant = camel4 INAP Language Extension ID=400 Scripter Audio Dir = ../../n2srp/demo/audio/ulaw Default Language = English SCP * Digit = B SCP # Digit = C
---------------	--	--	--

# Monitoring UI (cont.)

Tracing mode can be activated for Call Instances. Call Tracing Logs are stored in memory and can be accessed over the HTTP/S Monitoring UI.

The trace output shows:

- ❑ Protocol Messages In/Out.
- ❑ Debug/Dump-level Output.
- ❑ Warnings/Errors.
- ❑ Timestamps & Statistics.

```

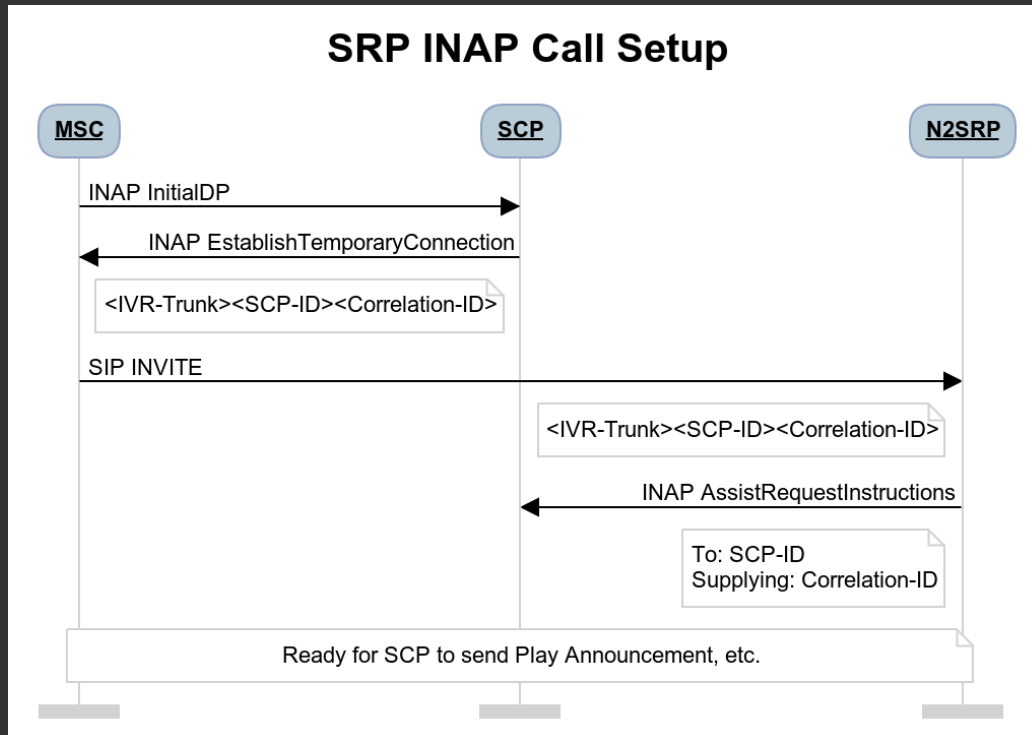
2018-06-05 16:55:54.503993 N2::Application [trace.debug] Incrementing internal stats 'instance.start'. Local value now = 1.
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] INVITE sip:9999211@10.42.2.251:5060 SIP/2.0
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] Via: SIP/2.0/UDP vagabond:5061;branch=z9hG4bK2073916970
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] From: sip:mss@vagabond;tag=2073622734
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] User-Agent: N-Squared MSS
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] Content-Length: 437
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] Content-Type: application/sdp
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] Call-ID: 2073947399@vagabond
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] Contact: sip:mss@vagabond
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] CSeq: 1 INVITE
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] To: sip:9999211@10.42.2.251:5060
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac]
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] v=0
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] o=jcouper 1949 2714 IN IP4 10.42.2.251
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] s=Talk
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] c=IN IP4 10.42.2.251
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] t=0 0
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] m=audio 7078 RTP/AVP 124 111 110 0 8 101
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=rtpmap:124 opus/48000
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=fmtp:124 useinbandfec=1; usedtx=1
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=rtpmap:111 speex/16000
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=fmtp:111 vbr=on
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=rtpmap:110 speex/8000
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=fmtp:110 vbr=on
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=rtpmap:101 telephone-event/8000
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=fmtp:101 0-11
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] m=video 9078 RTP/AVP 103 99
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=rtpmap:103 VP8/90000
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=rtpmap:99 MP4V-ES/90000
2018-06-05 16:55:54.504027 n2svcd [sip.srp_from_uac] a=fmtp:99 profile-level-id=3
2018-06-05 16:55:54.504233 SipApp::SipConnection [trace.debug] New SipServerTxn was sent by 10.42.2.251:5061.
2018-06-05 16:55:54.504343 SipApp::SipServerTxn [trace.debug] No rpt. Sent-By 'vagabond' will become explicit in Via 'received'.
2018-06-05 16:55:54.504397 SipApp::SipServerTxn [trace.debug] Handling Request Method for new INVITE Transaction.
2018-06-05 16:55:54.504446 SipApp::SipServerTxn [trace.debug] Initial INVITE From header (remote) tag = '2073622734'.
2018-06-05 16:55:54.504520 SrpApp::SrpInstance [trace.debug] Called Party '9999211' has CorrelationID (3 digits) = '921'. SCP ID (1 digits) = '1'.
2018-06-05 16:55:54.504689 SipApp::SipInstance [trace.debug] We need an RTP Worker now. Grab one from the pool.
2018-06-05 16:55:54.504729 SipApp::SipInstance [trace.debug] RTP [Worker-2] Assigning Session ID = 921, Session Version = 1528174554.
2018-06-05 16:55:54.504751 SipApp::SipInstance [trace.debug] RTP [Worker-2] Server Address [10.42.2.251:6970].
2018-06-05 16:55:54.504772 SipApp::SipInstance [trace.debug] RTP [Worker-2] Client Address [10.42.2.251:7078].
2018-06-05 16:55:54.504811 SipApp::SipInstance [trace.debug] RTP [Worker-2] INVITE SDP included telephone-event/8000 encoding id [101].
2018-06-05 16:55:54.504846 SipApp::SipInstance [trace.debug] Using 200 OK for DTMF Detection 'local'. Needs SDP 'a=sendrcv'.
2018-06-05 16:55:54.504865 SipApp::SipInstance [trace.debug] Requesting RTP worker to prepare. Will send INVITE response as soon as preparation is complete.
2018-06-05 16:55:54.504973 N2::Application::Worker [trace.debug] Application sending [4 + 176] bytes message to worker process.
2018-06-05 16:55:54.505084 SipApp::SipServerTxn [trace.debug] No response sent yet. Send 100 Result.
2018-06-05 16:55:54.505263 SipApp::SipServerTxn [trace.debug] SIP Server Transaction INVITE State Change <undef> -> SERVER:INVITE:PROCEEDING.
2018-06-05 16:55:54.505293 SipApp::SipServerTxn [trace.debug] Sending SIP INVITE Response Code 100 in State SERVER:INVITE:PROCEEDING.
2018-06-05 16:55:54.505313 SipApp::SipServerTxn [trace.debug] UDP Reply to Via Received Host '10.42.2.251', Via Sent-By Port = 5061 (or default).
2018-06-05 16:55:54.505345 SipApp::SipInstance [trace.debug] Skipping EDR Logging for A-Leg INVITE Response Code 100.
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] SIP/2.0 100 Trying
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] Via: SIP/2.0/UDP vagabond:5061;received=10.42.2.251;branch=z9hG4bK2073916970
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] From: sip:mss@vagabond;tag=2073622734
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] User-Agent: N-Squared SRP
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] Content-Length: 0
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] Call-ID: 2073947399@vagabond
2018-06-05 16:55:54.505462 n2svcd [sip.srp_to_uac] CSeq: 1 INVITE

```

# Message Flows

---

# Message Flows



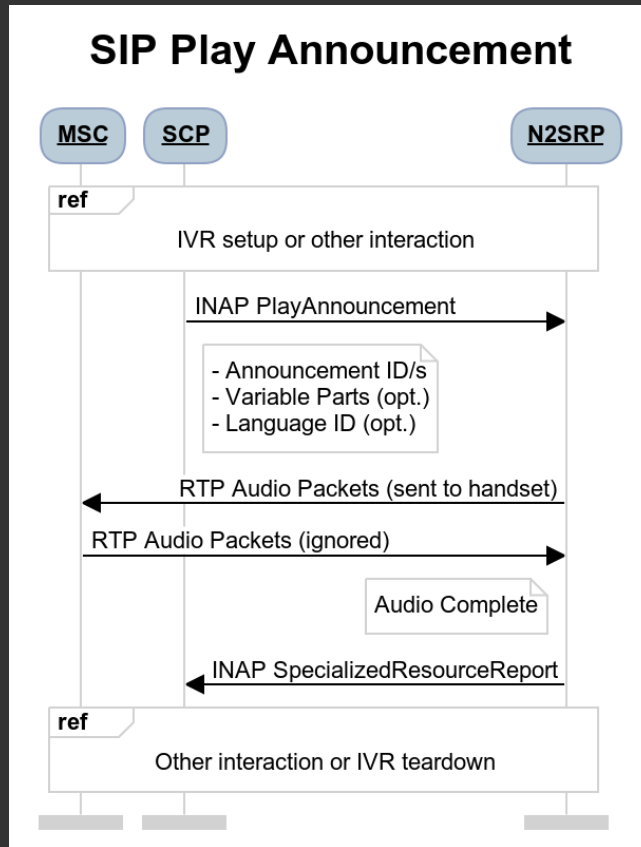
The Call is initiated by the MSC using INAP InitialDP to the SCP.

The SCP requests the MSC with EstablishTemporaryConnection to open a SIP channel to the SRP using SIP INVITE.

The SRP contacts the SCP to request further instructions using INAP AssistRequestInstructions.



# Message Flows (cont.)

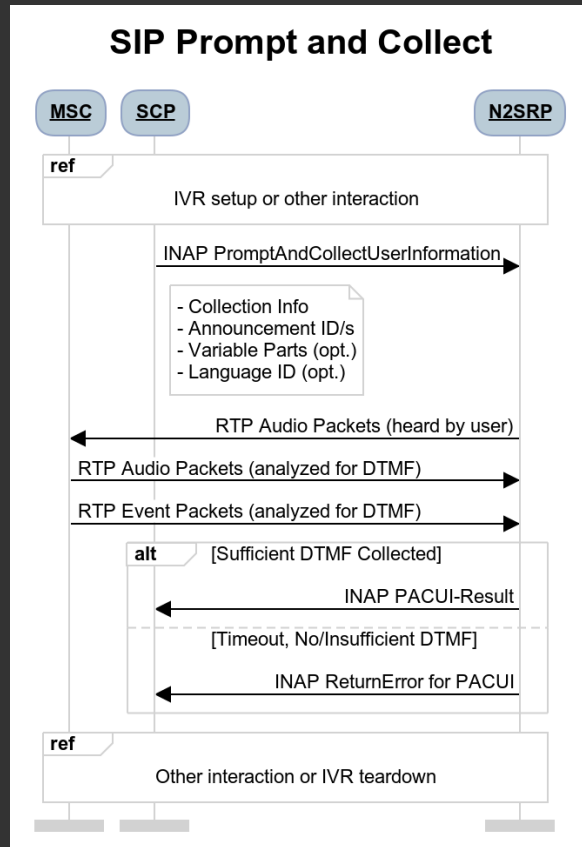


After the SRP has contacted the SCP, it can be instructed to Play Announcement over INAP.

The SRP concatenates audio fragments to construct the complete audio response.

Concatenation rules for numbers, dates, time and currency are different for each language.

# Message Flows (cont.)



A Prompt and Collect instruction from the SCP requires the SRP to perform DTMF detection.

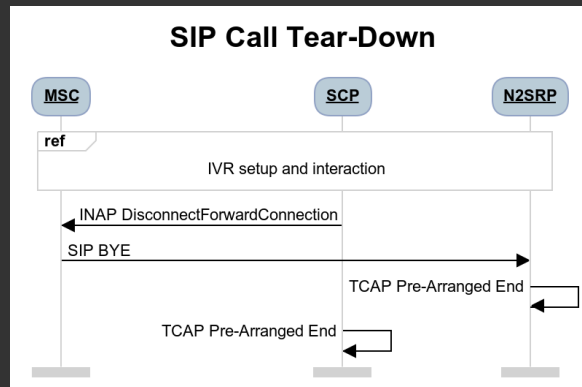
Some MSCs are capable of performing DTMF detection in hardware.

If the MSC does not support this, the SRP will use Fast Fourier Transform analysis in software.

# Message Flows (cont.)

The SCP is responsible for deciding when the SRP interaction session is over.

DisconnectForwardConnection is sent to the MSC, which uses SIP BYE to tear down the voice channel to the SRP node.



# Conclusion

---